

**EVEREST IST-2002-001858**

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*Final Report***Contractual Date of Delivery to the CEC: 31-12-2005****Actual Date of Delivery to the CEC: 15-01-2006****Author(s): Fernando Casadevall (UPC)****Participant(s): UPC, KCL, PTIN, TI, TID, TEL****Workpackage: WP1****Est. person months: 0.5****Security: PU****Nature: Report****Version: 0.1****Total number of pages: 49****Abstract:**

This deliverable constitutes the final report of the project IST-2002-001858 EVEREST. After its successful completion, the project presents this document that firstly summarizes the context, goal and the approach objective of the project. Then it presents a concise summary of the major goals and results, as well as highlights the most valuable lessons derived from the project work. A list of deliverables and publications is included in the annex.

For more detailed technical results please consider the public deliverables, available at <http://www.everest-ist.upc.es>

Keyword list: Co-operation with other Projects, Concertation Activities

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EXECUTIVE SUMMARY

This report summarises the main achievements of the EVEREST Project, an IST research and technological development project carried out between January 2004 and December 2005 by Universitat Politècnica de Catalunya (UPC); King's College London (KCL); Portugal Telecom Inovação (PTIN); Telecom Italia Lab (TILAB); Telefónica Investigación y Desarrollo (TID), TeliaSonera (TEL).

The most important technical achievements of the project cover many different aspects related to Radio Resource and QoS Management and Common Radio Resource Management (CRRM). Different algorithms related to Admission Control, Congestion Control as well as on Packet Scheduling procedures have been proposed and evaluated for the envisaged Radio Access Technologies, namely GERAN, UMTS, WLAN. Moreover issues related to the end-to-end QoS architecture have been also studied and evaluated.

The Performance evaluation of the proposed architecture and RRM/CRRM techniques was completed by means of a set of laboratory tests carried out using a Demonstrator developed in the project. This Demonstrator is a SW/HW flexible tool, which provides a realistic real time emulation of an evolved B3G radio access system able to manage multimedia IP based applications. Finally, significant dissemination policy, based on publications on high quality magazines and conferences, was carried out. Moreover, several standards contributions were also generated and presented to the pertinent 3GPP technical committees.

1. Overview of General Project Objectives

The provision of beyond 3G (B3G) heterogeneous network topologies is conceptually a very attractive notion; however, it is a challenge to accomplish an efficient network design. In this context, Radio Resource Management (RRM) strategies are responsible for an utmost efficient utilisation of the air interface resource in the available Radio Access Networks (RANs). Any stand-alone wireless systems or heterogeneous hybrids thereof, rely on RRM strategies to guarantee a certain prior agreed QoS.

The objective of EVEREST is to devise and assess **a set of specific strategies and algorithms** for access and core network, leading to an **optimal utilisation of scarcely available resources** for the support of mixed services with **end-to-end QoS mechanisms within heterogeneous networks B3G**.

In order to achieve the former main objective, the following partial objectives will be addressed in the project:

- ❖ To identify, propose, simulate, assess and validate **advanced RRM algorithms** for GERAN and UMTS as well as novel radio concepts as HSDPA and WLAN that represents the first evolutionary steps of current 3G systems toward beyond 3G.
- ❖ To develop **Common RRM** (CRRM) algorithms for heterogeneous networks focused on UTRAN and GERAN. Both tight and very tight coupling will be considered.
- ❖ To consider other technologies that can be complementary to GPRS/UMTS, such as:
 - WLAN for indoor hotspots
 - Different types of repeaters, acting as coverage extensions
- ❖ To investigate issues related to the **inter-working** of the core network (CN) Bandwidth Broker (BB) and the UMTS, GERAN and WLAN RRM entities (e.g., of how to connect the relevant elements in the two entities; what type of signalling to use; what mechanism and how often to update the information held in the BB; how to translate the parameters of RRM into the BB parameters; etc.).
- ❖ To demonstrate the benefits of the proposed RRM and CRRM algorithms by means of multimedia IP-based applications over a **real-time testbed**. In order to relate consistently network performance with advanced application usage, some examples of highly demanding services will be considered and related through the use of QoS management entities.

EVEREST is providing tangible contributions towards a heterogeneous realisation of 2G/2.5/3G (e.g., GERAN and UTRAN) and 3.5G networks (e.g., HSDPA) with the inclusion of newly emerging RANs (e.g., WLAN for hot spot coverage) for the 2007-2010 time-frame. Proposed solutions will be compliant with and aligned to standardisation activities in the field e.g. 3GPP, IETF, IEEE.

In order to accomplish the above objectives, the project evolves around to main activities: (1) algorithmic development and simulation by means of advanced simulation tools, and (2) demonstration of the technology by means of implementing real-time testbeds to prove concepts.

Research Challenges

Comparably little work has been devoted to date in providing solid and publicly available RRM strategies. This is due to the fact that RRM algorithms are not subject to standardisation, leading to an increase in competitiveness among the manufacturers. The aim of the project is to provide solid advances in this field, allowing for a truly optimised heterogeneous network deployment. The research challenges, to be tackled by EVEREST, can thus be summarised as follows:

- The RRM strategies of legacy networks (i.e. early releases of GERAN and UTRAN already being deployed by operators) are investigated. The GSM/GPRS part is of rather low dimensionality, i.e. only a few parameters are needed to tune their optimality. Joining these networks into a heterogeneous network, leverages the RRM strategies into optimisation problems with many more degrees of freedom.

- In the case of UMTS and UTRAN (UMTS Terrestrial RAN), a huge amount of human and material resources has been devoted to date that led to a vast plethora of 3GPP specifications, first manufacturer products and operator roll-outs. The focus to date has mainly been in radio network planning, since the first step was an initial network roll-out. Some very basic RRM strategies are being introduced on the presumption that the traffic would be low. In the coming years, when more and more radio engineers become familiar with 3G and beyond technologies, it would be necessary increase and harmonise the general knowledge on W-CDMA (i.e. UTRAN-FDD) RRM strategies. Only this will facilitate the future success of 3G and beyond networks with an ever-increasing sophistication in RRM strategies. Clearly, optimised RRM algorithms are the only key to successfully handle highly loaded networks, which are being envisaged to dominate the wireless arena in five years from now. Additionally, emerging concepts, such as RAN-sharing within UMTS will lead to mixed domains with equivalent public land mobile networks (ePLMN) solutions, complicating RRM strategies for specific services.
- WLANs are also expected to play an important role in the provision of high data rate services. Here, wireless indoor coverage could be better provided by WLAN as high power resources would have to be alternatively spent in UMTS. Again, a new dimension in the RRM problem is introduced, the influence of which needs to be assessed.
- For each technology, there will be a need to establish a mapping between equivalent bearers. This mapping will ease the definition of the inter-working from the service point of view.
- For each technology, the development of CRRM algorithms within the RAN is vital for a proper functioning of the heterogeneous network. Developed, assessed and compared have to be CRRM algorithms for tight and very tight coupling.
- Further, to support end-to-end QoS in a heterogeneous wireless (and wired) mobile environment, the interaction between the QoS management entities of the CN and the individual RRM of each RAN or the CRRM for a plethora of RANs, in an administrative domain is of prime importance. The CN of UMTS evolves towards an All-IP architecture with QoS mechanisms and protocols suggested by the IETF. Most of these mechanisms and protocols will be based on the BB concept. The BB is the main architecture element of the control plane of the Diffserv model proposed by IETF for supporting end-to-end QoS in IP-based networks. The BB is a logical entity responsible for resource allocation in an administrative domain and co-ordinates inter-domain and intra-domain resource allocation. For the inter-domain resource allocation, the BB of the CN should be aware of the resource requirements and resource availabilities of the peer CNs. For the intra-domain resource allocation, the BB of the CN should be aware of the resource availabilities of its underlying RANs.

The rational research approach endeavours to encompass the Radio Access Network (e.g. RRM for single RAT, CRRM algorithms), the Core Network (e.g. BB concepts for the envisaged architectures, Intra-CN-domain Diffserv Signalling Protocols, etc.) and a coupled architecture between both (e.g., comparative study on architecture and signalling issues; comparison between various CRRM + BB algorithms depending on loose, tight or very tight coupling).

Relevance of the objectives

EVEREST is framed within the "Mobile and Wireless Systems B3G" strategic objective (SO) of the IST priority. As stated in the objectives of this IST priority, "early preparatory work has characterised systems B3G as a horizontal communication model, where different terrestrial access levels and technologies are combined to complement each other in an optimum way for different service requirements and radio environments".

In this respect, the EVEREST research scope is fully aligned with what is known as systems B3G radio environments, since a diversity of technologies are considered: GERAN, UMTS and WLAN. EVEREST further stresses that early preparatory work has considered these combined and complementary scenarios to a very limited extent. Most of these technologies have thus not been sufficiently analysed to yield their full capabilities, e.g. QoS in GPRS; or, the studies have mainly focused on the first stage deployment capabilities, e.g. UMTS phase 1 capabilities and their limited associated features in terms of

RRM algorithms, implemented RABs, etc.; or, they have solely focused on the available product capabilities, e.g. IEEE 802.11b without QoS for WLAN.

Consequently, the goals of EVEREST include:

- Further progress on the definition of advanced RRM mechanisms (e.g. including for example location aided RRM algorithms) leading to an optimised usage of the different technologies according to a technology roadmap driven by the evolution of the wireless scenario;
- Acknowledge and contribute to a CRRM, where a pool of resources belonging to different technologies are commonly considered and commonly optimised.
- Use and test the CRRM, for providing end-to-end QoS in an IP mobile access network. Define the interactions between a BB and the radio entities, in order to provide the adapted QoS to the service and to use in an optimal way the heterogeneity of the IP access network.

This project is clearly network operator-driven. They have identified the scenarios of interest to be considered at different time scales and for different network roll-out phases. The close involvement of Mobile Operators guarantees that realistic network data (e.g. traffic distributions, propagation losses, data traces, etc.) can be processed and utilised to the benefit of optimised RRM algorithms for heterogeneous networks. From the operator side, it is well understood that suitable approaches for RRM fall well beyond a mere network deployment; they constitute an innovative research field with clear indications on how to manage radio resources in heterogeneous networks accommodating traditional and novel services. In this respect, the operator-oriented approach adopted in EVEREST ensures that the analysed scenarios are market-relevant and user-centric.

Moreover, results coming from the project will provide a manufacturer-independent and complementary analysis of the RRM strategies. This will allow the mobile operators to evaluate and compare solutions coming from the market with an available reference of the system performance. It is worth noting that the open nature of the algorithms developed within the project and its availability to the entire wireless community is expected to contribute to a better transition from the different evolutionary scenarios considered. Outputs from EVEREST will constitute a valuable reference for operators, manufacturers and academia, facilitating further progress in this field for many years to come.

2.- Contractor Involved

Participant Name	Short Name	Country	URL Address
Universitat Politècnica de Catalunya	UPC	Spain	http://www.upc.edu
King's College London	KCL	United Kingdom	http://www.kcl.ac.uk
Portugal Telecom Inovação	PTIN	Portugal	http://www.ptinovacao.pt
Telecom. Italia	TI	Italy	http://www.telecomitalialab.com
TELEFONICA I+D	TID	Spain	http://www.tid.es
TELIASONERA	TEL	Sweden	http://www.teliasonera.se

3.- *Work Performed*

To cope with the above mentioned objectives, EVEREST is being developed considering the following three main stages:

1. Determination of interest and relevant target scenarios. This takes into account, the communications environment (macrocell, microcell, indoor, etc.), and user mobility, the technologies deployed (GSM, GPRS, EDGE, UMTS, WLAN), the network architecture, as well as the services (conversational, interactive, streaming, etc.).
2. Development of RRM and QoS management algorithms, with evaluation through simulation. Focus is being placed on finding commonalities among the different scenarios considered, rather than trying to optimise algorithms and algorithmic parameters for a specific scenario. Thus, the goals of EVEREST extend the mere analysis of different scenarios and it targets the definition of generic RRM criteria, facilitating their applicability in scenarios differing from those studied in detail within the project.
3. Validation and demonstration of the proposed algorithms for the defined scenarios by means of a real time testbed supporting IP-based mobile multimedia applications with end-to-end QoS capabilities. To support this latter service, the IP CN has to be configured in the following way: a mobility management is installed, and the QoS framework based on Diffserv, including its control plane, is configured. Then, the interactions of the BB with the radio resource entities are considered.

These main research topics in EVEREST are addressed within a proposed **End-to-End QoS management** framework aligned as much as possible with the QoS architecture envisaged in 3GPP Release 5 and 6 and other relevant IETF proposals. In this sense, it is assumed within the project that any end-to-end QoS architecture for converged 3G mobile – wired IP networks should be compliant with 3GPP UMTS QoS general framework (ref. 3GPP TS 23.107, TS 23.207). In particular, a QoS management architecture extending the 3GPP policy-based concepts is considered, and the following issues are envisaged to fulfil B3G QoS requirements:

- Introduce E2E resource based admission control mechanisms.
- Extend the policy-based framework to cope with resource management in the radio access part as well as in the CN.
- Use policy technologies to manage the derivation of UMTS QoS parameters.

Figure 1 illustrates the proposed architecture over a B3G network where different RANs can be offered to access at the same Core Network. The key aspects of this QoS management architecture are the following:

- The PDF function already introduced in 3GPP R5/R6 policy framework is maintained and two new entities are introduced in the B3G QoS architecture, namely: the Bandwidth Broker (BB) and the wireless QoS broker (WQB). The BB is in charge of the control plane of the DiffServ domain, while the WQB is the counterpart of the BB for the radio part of the access network. A clear parallelism can be done between the WQB and the BB.
- Both BB and WQB could act as policy managers and their policies are enforced in the core network routers as well as in the radio equipment respectively. Common RRM strategies are managed by the WQB.

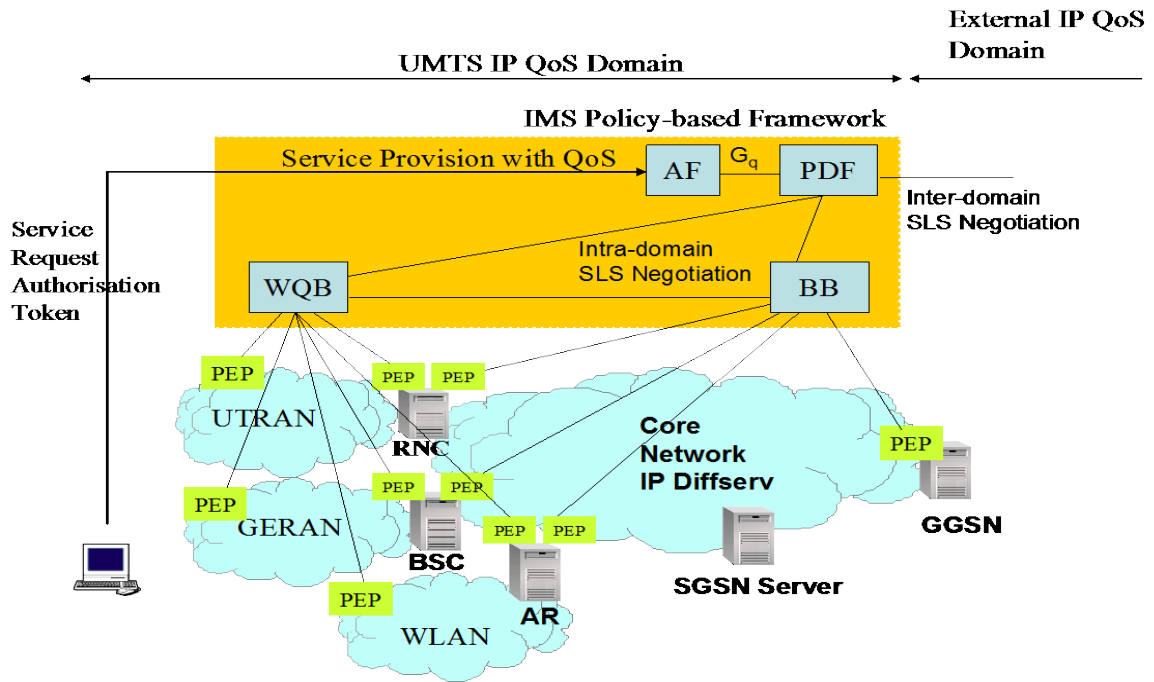


Figure 1. -EVEREST QoS architecture in a heterogeneous radio access network.

- The relationship between the PDF entity from IMS and the new entities WQB and BB is envisaged in terms of QoS negotiation. Furthermore this QoS negotiation is extended from the PDF, which really acts as a master PDP of the domain, towards external neighbouring domains.
- The proposed architecture is valid for any degree of coupling among the heterogeneous RANs although the commitment of the WQB and CRRM functions would effectively depend on the degree of coupling.
- SGSN control functions are separated from its routing functions. This approach is intended to introduce native IP transport down to the RNC and its equivalents.

Testbed

The EVEREST testbed is aimed at demonstrating most of the concepts addressed within the project and analyse them under the set of scenarios identified. The testbed builds a GERAN/UMTS/WLAN stand-alone real time emulator platform, including all the relevant QoS entities in both the radio access part and the CN, to show and analyse the end-to-end QoS performance. Such approach will allow testing multimedia IP-based applications (videoconference, streaming services, web browsing, etc.) on an end-to end basis and over an emulated access network with enhanced RRM features. Among other EVEREST testbed features, it should be emphasised:

- To test the end-to-end Quality-of-Service (QoS) performance.
- To assess, in real time, the effects that RRM/CRRM/BB algorithms have on the user's perceived QoS.
- To verify the following identified procedures: *Initial RAT Selection; RAT Switching, Connection establishment with E2E QoS negotiation, E2E QoS Re-negotiation, CN Mobility Management and QoS interactions, Common Radio Resource Management operation, Impact of CRRM and QoS Management on Applications.*

The architecture proposed for the EVEREST testbed reproduces the reference architecture depicted in figure 1. First of all, a cluster of PC's, devoted to perform the emulation of the Heterogeneous Radio Access Network, could be identified. Next, a second group of PC's together with other network elements (routers) deal with the emulation of the UMTS Core Network. The third element envisaged here could be the external IP backbone, and finally the PCs working as user terminal and server of applications.

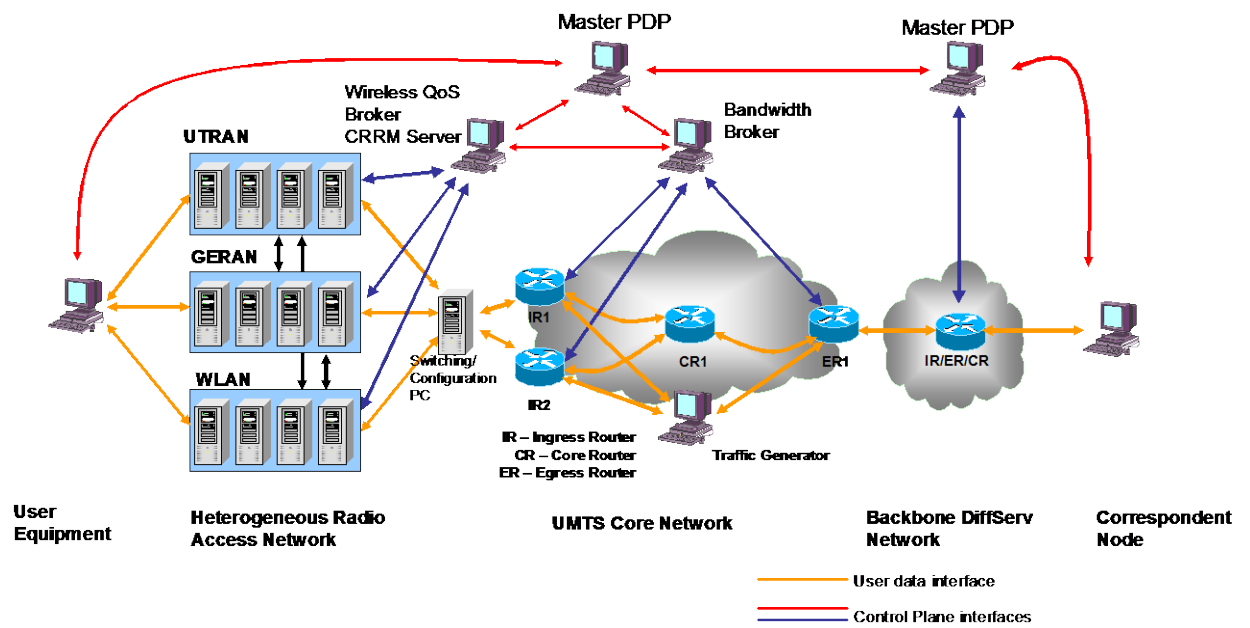


Figure 2. -General architecture of the EVEREST Testbed

4.- End Results

During the development of the EVEREST project the following main results have been achieved:

4.1 New E2E QoS architecture

A policy-based framework is introduced in 3GPP release 5 to manage QoS for multimedia services supported within the IMS Domain and the framework is extended to other services in release 6. This policy framework is intended to enable the coordination between events at the application/service layer and resource management at the IP bearer layer and it can be used to provide a policy-based admission control in charge of authorizing specific QoS resources for a set of IP flows within a user session. In this way, the service provider (e.g. the mobile operator) could decide which level of IP QoS is offered taking into consideration the characteristics of the service being requested but also any other consideration related to business models and management (premium users, etc.).

The policy-based UMTS framework introduces the Policy Decision Function (PDF) entity and the Policy Enforcement Point (PEP), located in this case in the GGSN node of the Core Network. Additionally, the policy-based framework in release 6 considers the existence of a logical element referred to as Application Function (AF), which is really in charge of offering services that require the control of IP bearer resources. The AF entity for IMS services (based on Session Initiation Protocol, SIP) is the SIP proxy referred to as P-CSCF (Proxy Call Session Control Function) in 3GPP terminology. In release 5 the PDF is collocated with the AF. A simplified view of this policy framework is illustrated in Figure 3.

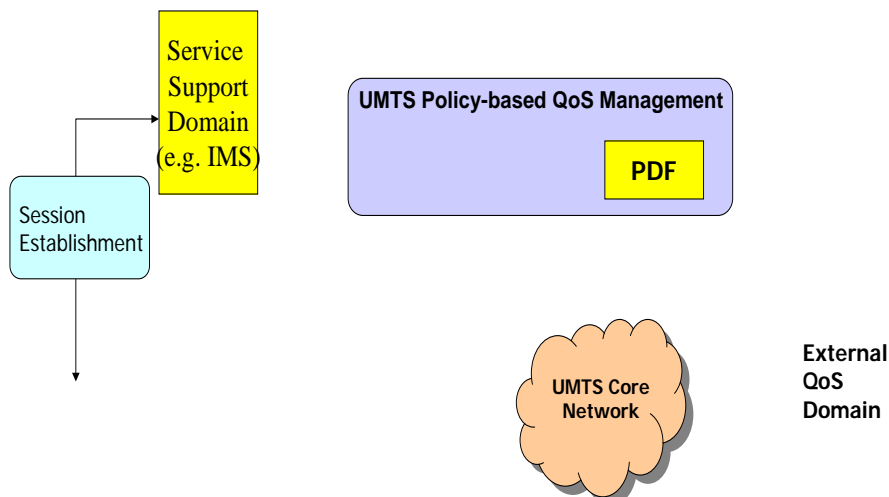


Figure 3.- 3GPP policy-based QoS framework

The UMTS policy-based QoS framework is mainly addressed to provide authorised IP QoS parameters for IMS sessions at the GGSN so that these IP QoS parameters can be enforced to the UMTS network by using translation functions to obtain the correspondent UMTS bearer attributes. Nevertheless, it is worthy to remark that QoS control inside the UMTS network for internal bearer management (i.e. radio access and core network bearer services) is completely unaware of this policy-based framework.

A QoS management architecture extending the 3GPP policy-based framework is proposed in EVEREST to address the QoS problematic within B3G networks. The 3GPP policy-based framework for IMS services can be seen as the first step to introduce policy-based technologies in wireless networks but several extensions would be required to fulfil B3G QoS requirements:

- In the current 3GPP QoS framework, authorization of a session in the PDF is done without taking into account radio network resources. This could lead to situations in which a session is authorized at the PDF but then the admission control mechanism triggered by the PDP Context activation may reject the establishment of resources for that request. Although in a single access network scenario such as UMTS this situation does not have too much relevance, in a scenario where several access network

options co-exist this situation could lead to block a session authorized over a specific RAT (e.g. selected based on operator preferences) while enough resources are available in another one.

- Allow end-to-end resource-based admission control. From the same reasons stated in previous paragraph, the PDF should authorize sessions not only based on local domain policies but addressing resource limitations or congestion in the external network(s).
- The policy-based framework should be extended to cope with resource management in the radio access part as well as in the CN. In this sense, resource usage in the entire B3G network elements is expected to operate under a set of policies that guides the system behaviour. In particular, a policy-based RAT selection decision function is considered mandatory within the QoS management framework as well as the introduction of dynamic QoS negotiation mechanisms (ie. dynamic traffic class mapping) among all the potential RATs and the CN. As stated in previous section, in the current 3GPP solution policy-based management is limited to IP QoS resource authorisation at the GGSN.
- QoS management should encompass Common Radio Resource Management (CRRM) in coordinated RANs connected to the same CN. So, CRRM mechanisms should be made available somehow to the QoS management framework.

Figure 4 illustrates the main concepts of the envisaged QoS architecture over a B3G network.

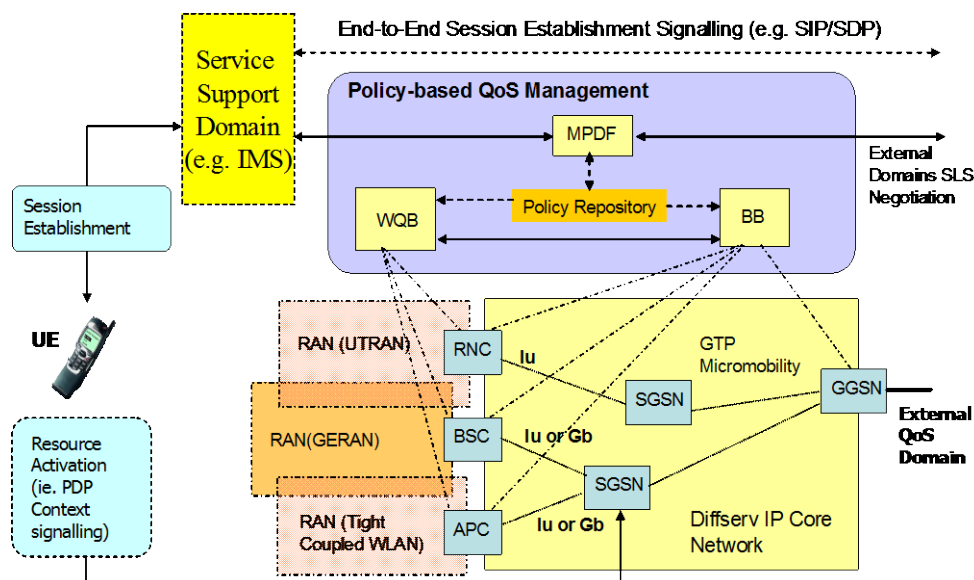


Figure 4.-. Reference B3G network UMTS-based architecture

A DiffServ enabled IP network is assumed to be used in the core network. The key aspects of this QoS management architecture are the following:

- Two new functional entities are introduced to support the policy-based approach: the Bandwidth Broker (BB) and the Wireless QoS broker (WQB). The BB is in charge of the control plane of the DiffServ domain while the WQB is the counterpart of the BB for the radio part of the access network. A clear parallelism can be done between the WQB and the BB. It is worth noting that the Policy Decision Function (PDF) already introduced in 3GPP R5/R6 policy framework is maintained. Nevertheless, it is referred here as Master PDF (MPDF) in Figure 4.
- Relationship between the M-PDF and the new entities WQB and BB is envisaged in terms of QoS negotiation. Then, session's QoS-requirements for the whole B3G network domain are provisioned accordingly in the radio access and in the core network as a result of this negotiation. Furthermore, the master PDF is in charge of QoS negotiation with external peer domains involved in the provisioning of end-to-end services. Inter-domain QoS negotiation can follow different approaches. For instance, network wide policies could be established among peer domains through the exchange of updated Service Level Agreement (SLA) information.
- Negotiation of QoS between the user (or a proxy in the Service Support Domain) and the MPDF is achieved by a policy-based service negotiation protocol (e.g. COPS-SLS). What differentiate policy-

based negotiation protocols are the flexibility of the signalling and the independence of the signalling from the QoS mechanism used in each domain. Each domain might have a specific set of negotiation parameters (or a set of Service Level Specifications, SLSSs) and specific policy for admission control.

- o The proposed architecture is valid for any degree of coupling among the heterogeneous RANs. For instance, Figure 4 shows a tight coupling solution where the WLAN functionality is seen as a UTRAN/GERAN network and as such an Iu-PS interface would be used to connect it to an SGSN. Notice that, the role of the core network is to allow the exchange of information between the radio network controllers of these RANs, thus making possible the use of common radio resource management strategies. In such case, in order to support RRM and CRRM functionalities, a new element is required in the WLAN, referred to as Access Point Controller (APC), with equivalent functions like the RNC or the BSC for the UTRAN and GERAN. In particular the APC architecture design could follow the Generic Access Network (GAN) approach¹.

QoS Management in RANs: WQB concept

The Wireless QoS Broker entity can be seen as the counterpart of the BB for the radio part of the access network although functions deployed may have a completely different approach due to the specific characteristics of the radio access part. The envisaged functions for the Wireless QoS Broker entity are represented in Figure 5 and described hereafter:

- Configuration of RAN elements for QoS provision. One function of the WQB is to act as a PDP in order to configure QoS mechanisms available at the RANs. As each RAN may have specific QoS mechanisms, the WQB is responsible of the configuration of such mechanisms in order to achieve the expected behaviour.
- Common Radio Resource Management (CRRM) is able to steer the traffic distribution among the RATs towards an optimal distribution increasing the radio resource efficiency and improving the perceived service quality. Then, in the envisaged architecture, the WQB entity would allocate two main functionalities related to CRRM: coordination of the different access resource pools, each with their own RRM functionalities, and selection of the most suitable RAT through call admission control or inter-system handover procedures in terms of some operator defined policies.
- Intra-domain QoS negotiation. Coordination is needed between the WQB and the BB, as the admission control and handover decision are submitted to different constraints in the radio part and the IP CN of the mobile access network. In the first case these constraints are related to the radio resource usage, and in the latter case to the network topology and the traffic distribution in the network.

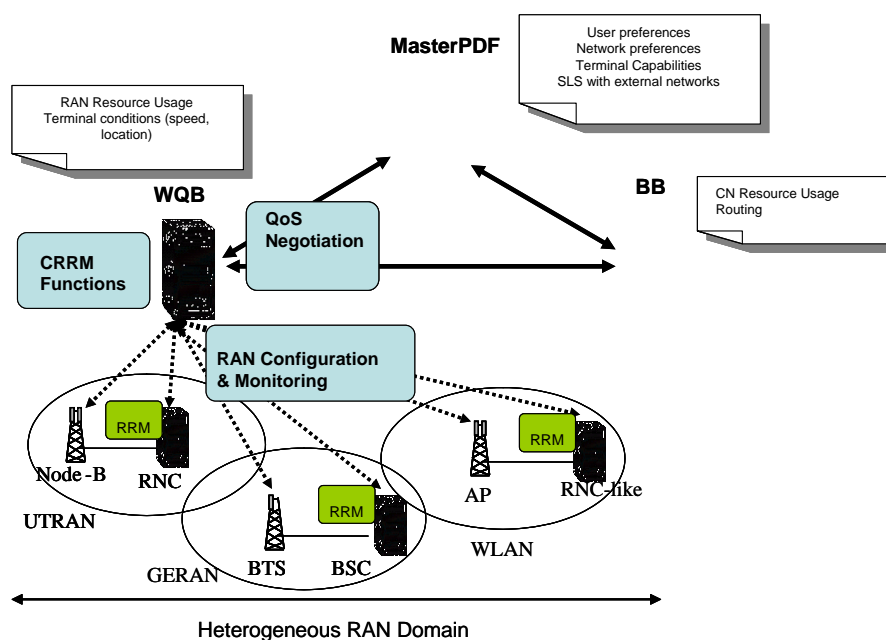


Figure 5. Envisaged functions for the proposed Wireless QoS Manager

¹ A generic access network (GAN) is a broadband IP network providing access to A/Gb interfaces of GERAN/UTRAN Core Network

In summary, a new end-to-end (E2E) QoS architecture, including message sequence chart for the End-to-End QoS signalling, has been proposed. The proposal follows a policy-based approach. A new concept, Wireless QoS Broker (WQB), has been introduced for managing the resources of the radio access part within the framework of B3G networks. The architecture presented encompasses CRRM functionality support in the radio part and proposes a QoS control plane to jointly manage IP resources in the core network and radio bearers in the RANs. The proposed architecture was also developed in terms of procedures and signalling and QoS class mapping between UMTS and DiffServ classes.

4.2 Evaluation of RRM/CRRM algorithms

RRM is a complex problem with many factors influencing in the achieved performance and with many mixing effects. Furthermore, the RRM problem has multiple dimensions and multiple functionalities that, either in a more direct or indirect way, impact on the air interface. Then, in the case of EVEREST, the RRM mechanisms considered are: GERAN, UTRAN and WLAN

RRM issues for UMTS

WCDMA access networks, such as the considered in UTRA-FDD proposal, provide an inherent flexibility to handle the provision of future 3G mobile multimedia services. UMTS will offer an optimization of capacity in the air interface by means of efficient Radio Resource Management (RRM) algorithms. RRM is a complex problem with many factors influencing in the achieved performance and with many mixing effects. Then, a crucial aspect is to identify relevant issues and fundamental elements influencing on the overall RRM process, then achieving a wide-scope and open-minded perspective. In this context, the subsections detailed below describe a variety of studies related to RRM for UMTS.

I- A new framework for capturing coupling among cells

An innovative mathematical framework capturing the air interface coupling among the different cells in the scenario has been developed based on the derivatives of the cell uplink load factor and the downlink transmitted power [1][2]. This framework is presented in a compact formulation for both uplink and downlink, and it allows implementing mechanisms supporting smart load control actions including admission control, congestion control algorithms. For instance, figures 6 and 7 show how the algorithm based on derivatives achieves the load reduction with the lowest NRT throughput reduction. The difference with respect to an algorithm that selects in a random way ("aleat" algorithm) the NRT user whose throughput should be decreased is very high. On the contrary, both algorithms based on derivatives or even in interactive load perform much better.

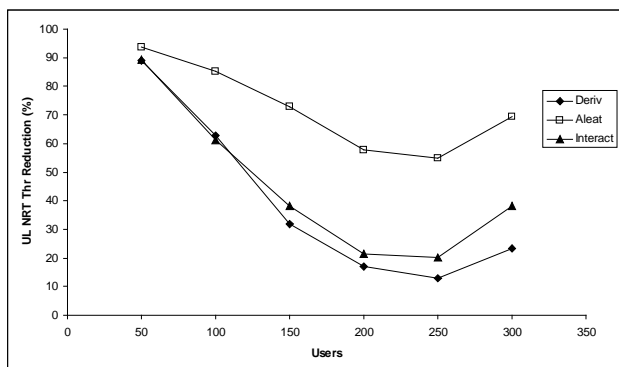


Figure 6.- UL NRT Throughput reduction in the neighbouring cells

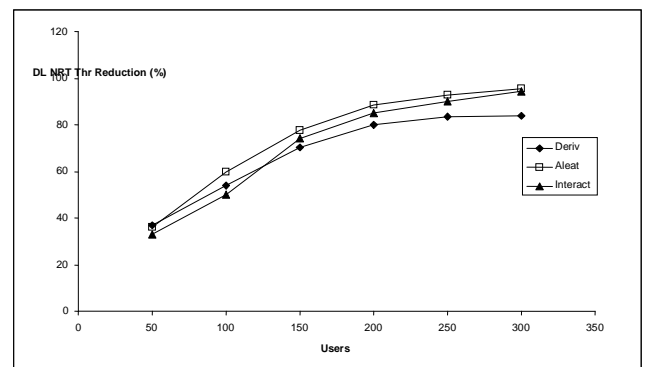


Figure 7.- DL NRT Throughput reduction in the neighbouring cells

II- Indoor Traffic

The implications of indoor traffic in 3G W-CDMA based systems may significantly differ from 2G TDMA-based solutions because transmitted power levels are the key radio resources in W-CDMA. The higher power levels needed for indoor service will lower cell capacity both for uplink and downlink traffic. Table 1 presents the capacity loss for both uplink and downlink as the fraction of indoor traffic increases. Capacity is defined as the maximum number of users in the scenario that guarantees a BLER $\leq 2\%$. It can be observed that the degradation due to indoor users is much more significant in the uplink than in the downlink direction.

Table 1 .- Capacity loss (%) relative to the case with no indoor traffic ($p=0$) for 64/64 kb/s and 64/384 kb/s radio bearer

	UL 64 Kb/s	DL 64 Kb/s	DL 384 Kb/s
P=0.1	19.2%	9.3%	12.5%
P=0.2	37.7%	11.6%	18.8%
P=0.5	88.4%	15.3%	25.0%

III- Traffic Hot-Spots

In a real mobile network, there are certain geographical areas with high density of users. In these areas, a high demand of radio resources may appear. In order to assure the user QoS (Quality of Service) requirements in a hotspot, not only network planning but also RRM algorithms must be considered.

Figure 8 shows the impact of a hotspot (HS) location on the transmission power of two base stations, named 5 and 10 respectively. As shown, when the hotspot is located far from Base Station 5 (HS at 200m), the power increase is higher than when the hotspot is nearer (HS at 50m). This is because hotspot users located far from the base station demand higher level of power to guarantee the (E_b/N_o) target. So, if the hotspot is located relatively far from its base station, it can happen that the Base Station may not have enough power to satisfy all user requirements, causing bad signal quality. A connection is dropped if the current (E_b/N_o) is 1dB below the target value during the 90% of time in 1second.

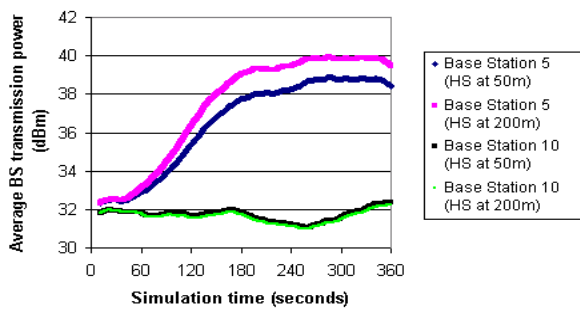


Figure 8.- Effect of static hotspot location on BS power

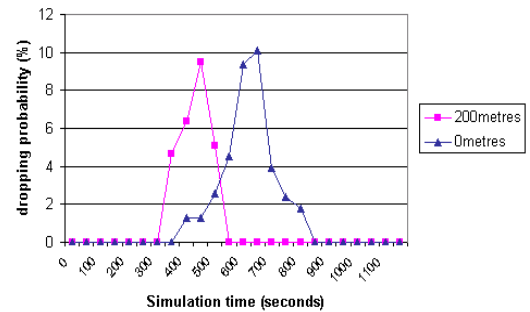


Figure 9.- Impact of hotspot movement on dropping probability

It is worth noting that if the hotspot users were initially located at 0metres from BS5, the overload situation in BS5 would last longer because the hotspot users connected to BS5 would need much time to reach the coverage area of BS10 and make handovers. When hotspot users arrive near the cell edge, they will begin to handover to BS 10, and then, the BS 5 transmission power will be reduced. Obviously, this fact will happen first if the hotspot is initially located at 200m from BS. Moreover, when the hotspot is initially located near BS5 the overload situation and the droppings occur later, and then the available time to prevent this overload is higher such as it is shown in figure 9.

In the uplink case, the load factor of different base stations and the system performance in terms of dropping probability have been analysed and similar results to those obtained for the downlink are reached.

Another technique that is commonly used is to adjust the transmission pilot power of the hotspot cells and its adjacent cells [3],[4]. Figure 10 compares the transmission power of BS5 and BS10 with the proposed pilot adjustment algorithm (with load balancing) and without the proposed pilot adjustment algorithm. The pilot power is changed once in a second ($T_p=1$ second) and $\alpha=1$. The parameter α determines the sensitivity in the changes of the pilot power. As shown, when no load balancing is considered, an overload of BS5 and BS10 can be observed as users move from BS5 to BS10. With the load balancing technique, the transmission power of BS5 and BS10 is equalised (see figure 10) adjusting the pilot in an adequate way as shown in figure 11, where the average value of the pilot power of different BS is plotted. The load balancing technique reduces the power limitation probability of the different base stations, and this will reduce the dropping probability as it is shown in Table 2.

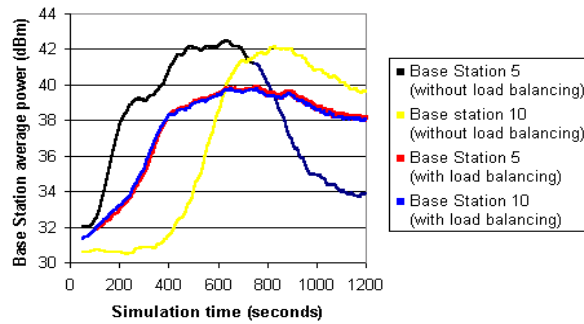


Figure 10.- Impact of load balancing in BS transmission power

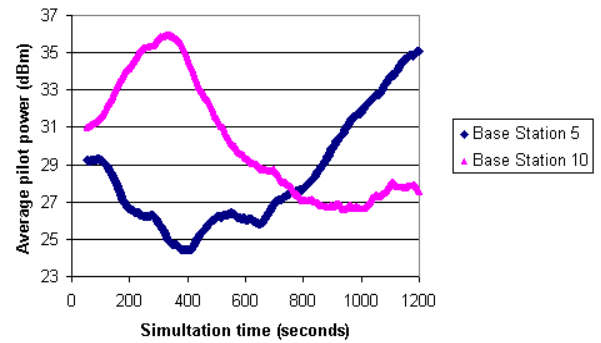


Figure 11.- Evolution of the pilot power of different BS when the proposed pilot adjustment algorithm is applied

On the other hand Table 3 also shows the dropping probability and the average time between handovers per user, for different values of α and when no load balancing is carried out. Once again, the proposed algorithm reduces the dropping probability with respect to the non-load balancing case, but it increases the rate of handovers causing an increase in network signalling. Moreover, the most appropriate value of α is between 0.1 and 0.25.

Table 2.- Impact of T_p on dropping probability and handover statistics

	No Balance	$T_p = 1$	$T_p = 10$
Dropping prob.	2.38	0.51	0.73
Avg. Time between HO per user (sec.)	42.93	12.24	18.81

Table 3.- Impact of α in dropping and handover statistics.

	No Balance	$\alpha = 0.1$	$\alpha = 0.25$	$\alpha = 1$
Dropping prob.	2.38	0.16	0.14	0.51
Avg. Time between HO per user (sec.)	42.93	34.71	23.47	12.24

IV- Static Traffic

The flexibility in the provision of multiple bit rate services in 3G communication systems will allow users to benefit from services better adjusted to their specific requirements. Higher bit rates, QoS (Quality of Service) will also be crucial for 3G success from the user point of view. In this context, it becomes prime important to identify the key elements characterising the different services and anticipate the required mechanisms to support these services through the air interface in a suitable and optimised manner. In particular, users that receive data traffic services, typically with laptops in scenarios like offices, airports, etc., use to be static or, at least, with a very limited mobility. This fact gives room to propose more sophisticated admission control strategies, which may provide significant performance improvements.

Under this framework, an algorithm, denoted as PLEBAC (Path Loss Estimation Based Admission Control) has been proposed taking into account the benefits from the easier predictability in terms of power consumption of static data users. It makes use of the measurement reports provided by the terminal during the call set-up process in order to have a more accurate estimation of the required power along the connection time. The proposed algorithm has been evaluated under different conditions of service bit rate and cell radii and compared against a reference algorithm and compared against an algorithm that simply considers the average power consumption, denoted as PABAC (Power Averaged Based Admission Control).

In order to assess the potential of the proposed PLEBAC algorithm, a set of system level simulations have been carried out. Different scenarios have been considered with different cell radii, ranging from 500 m up to 2 Km. Figure 12 presents the transmitted power and the power increase estimation for both algorithms as a function of the path loss. Notice that PLEBAC adjusts more accurately the power increase estimation to the real transmitted power, which is higher when the path loss increases.

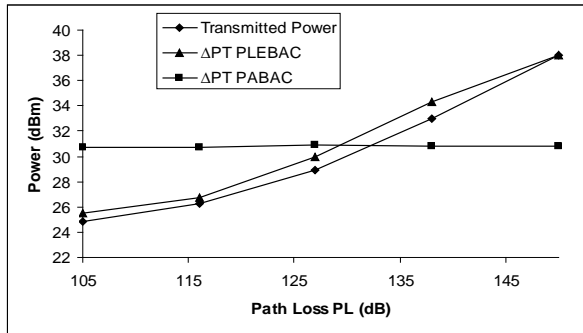


Figure 12 .- Actual transmitted power and power increase as a function of the path loss

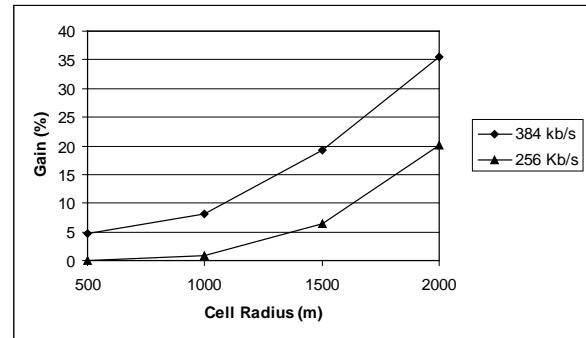


Figure 13 .- Throughput gain as a function of the cell radius

Figure 13 illustrates the throughput gain of the PLEBAC admission control with respect to PABAC for different cell radii, when the offered bit rate is 256 kb/s and 384 kb/s. Notice that the gain is bigger for larger cell radii and higher service bit rates. In these cases, the higher power demand of users with larger path loss leads to performance degradation not only for these users but also for other users that are located closer to the base station. As a result, an algorithm like PLEBAC, which takes into account user's path loss, improves the overall performance while at the same time it guarantees the quality of the accepted users. Furthermore, the higher the service bit rate, the higher the contribution of a user to the total system throughput, and consequently, a bad admission or a bad rejection turns into larger throughput reductions. As a result, it can be concluded that the PLEBAC strategy is better adapted to user's distribution in the network.

V- Repeaters

The repeaters are equipments between the radio base station and the terminals, able to amplify the received signals both on the uplink and on the downlink. Typically, a repeater is formed by two antennas: one antenna (called donor antenna) is strongly directive and is directed to the donor base station; the other antenna (called coverage or service antenna) has the purpose to cover the service area. The possible usage situations for repeaters in a WCDMA system are the following:

- coverage extension: in order to cover the so called "dead spot" (areas not covered during the first deployment of the network);
- capacity extension: in order to increase the capacity of a base station with an increased traffic load;
- soft-handover region reduction: thanks to the repeaters it is possible to reduce the soft-handover areas for already-connected users.

When a repeater is introduced in a WCDMA system, the first effect that it is possible to observe is the increase of the noise figure of the base station. Considering the coverage area, with the increase of the noise figure of the base station we have the receiver sensitivity making worse, with the consequence that the base station coverage decreases, although the total cell radius is increased due to the introduction of the repeater.

In terms of capacity, the introduction of the repeaters leads to a modification of the interference characteristics of the scenario. Figure 14 shows the percentage of served users when varying the repeater gain. Two load situations: 50 users per cell and 20 users per hot-spot; 60 users per cell and 25 users per hot-spot have been considered and three different conditions have been analyzed:

- Neglecting the effects of the repeaters on the noise figure and the blocking due to codes in downlink (case III)
- Considering the effects of the repeaters on the noise figure and neglecting the blocking due to codes in downlink (case II)
- Considering both the effects of the repeaters on the noise figure and the blocking due to codes in downlink (case I).

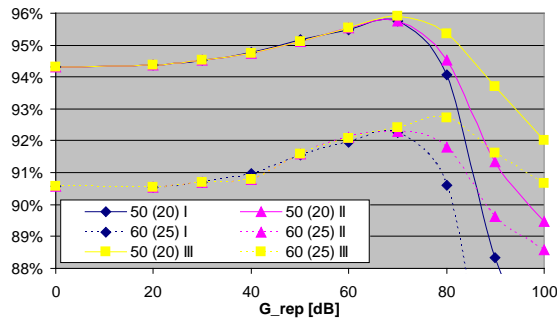


Figure 14 – Percentage of served users (I=noise figure modified, blocking on codes in downlink; II=noise figure modified, no blocking on codes; III=no noise figure modified, no blocking on codes).

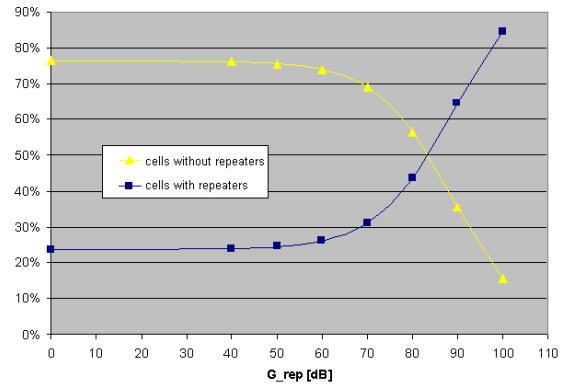


Figure 15 – Best server area variation in cells with and without repeaters

The percentage of served users in the system increases with repeater gain values lower than 70-80 dB, when the maximum capacity value is obtained. Certainly, increasing the repeater gain, the influence of the repeater on the surrounding area increases (see figure 15) and therefore the *best server area* of its donor base station increases also. Thus, in the hot-spot area the out-of-service users decrease and, at the same time, the number of users transmitting to the donor cell through the repeater increases.

VI- Multiple RF Carriers

The scenario simulated consist in an indoor traffic Hot Spot within an urban area. The UMTS operator uses two frequencies, where frequency f2 only applies at a three sector site and frequency f1 is applied at all the macro layer of cells. The study considers a mix of CS (speech) and PS (HTTP) traffic. Of the offered load in Bytes 25% is speech and 75% is HTTP. Different methods affecting on blocking ratio, dropping ratio and bit rate have been studied. The methods considered are unsymmetrical allocation of load to frequency f1 and f2, redirection of blocked service requests on frequency f1 to frequency f2 (or vice versa), allocation of all speech to one frequency and turning off soft handover on frequency f2. As a reference method 50% of the speech requests are allocated to frequency f1 and the rest of the speech requests are allocated to frequency f2. The same allocation probability is applied for HTTP.

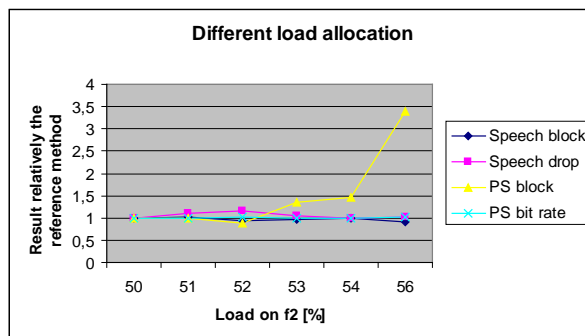


Figure 16. - 50 to 56 % of the load, speech and HTTP, is allocated to frequency f2.

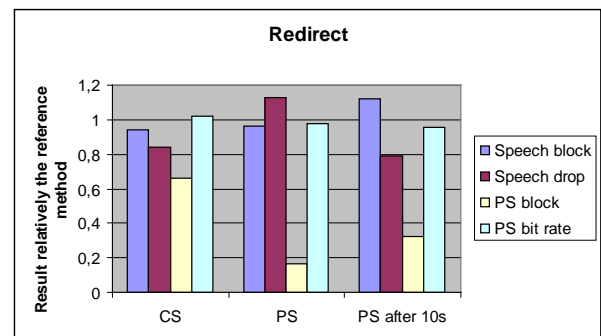


Figure 17.- Result obtain when redirecting service requests blocked on frequency f1 to f2, and viceversa.

Figure 16 below shows the impact of allocating more and more speech and HTTP load to frequency f2. We see that it is the PS blocking that is affected mostly as the load increases. The PS blocking is as smallest when 52% of the load is allocated to frequency f2, but it starts to increase if f2 is loaded with more load. The speech blocking and dropping is fairly insensitive to shifting more and more load from frequency f1 to f2 (at least in the load range tested here). Hence, the admission control method protects the speech users from deteriorating QoS when load increases, as it should. The PS bit rate does not change much either as f2 carries more load. The results indicate that PS blocking is sensitive to the load allocation on the frequencies, whereas speech blocking and dropping, and PS bit rate, is less sensitive to different load distributions. HTTP blocking is improved and the speech QoS is worsening when all speech is allocated to frequency f1. Speech QoS is improved then all speech is allocated to frequency f2 (assuming that inter-frequency handover does not cause dropped calls). If soft handover is

tuned off on frequency f2, the PS bit rate is increased, but, unfortunately, the PS blocking is also increased.

The result obtained when redirecting service requests blocked on frequency f1 to f2, and vice versa, is shown in Figure 17. The first result in Figure 17, called “CS”, is the result of redirecting speech service requests. The second result, called “PS”, is the result of redirecting HTTP service requests when ever they are not admitted on the serving frequency. The third result, “PS after 10s”, is when HTTP service requests are redirected first after been queued 10 seconds on the serving frequency, i.e., after the HTTP service request has registered a PS block on the current frequency. We see in the result that both speech and HTTP gains when redirecting speech services. The speech blocking is however only reduced about 6%, which is fairly little. On the other hand, when redirecting PS services, the PS blocking is reduced 84%. The speech service does not improve its QoS when HTTP requests are redirected. When redirecting a HTTP service first after 10 seconds, the PS blocking is also improved a lot. For this case the PS blocking is reduced 68%.

VII- Hierarchical Cell Structures

In this section some dynamic evaluations on Hierarchical Cell Structure (HCS), taking into account macro and micro layers, are reported. The activity's goals were to identify how to set the thresholds that control the algorithm applied by the terminal for the identification of the high/low mobility state by means of the number of cell reselections performed in a fixed period of time.

From Figures 18 and 19 it could be noted that, decreasing the Q_{offset}^2 values of macro cells, the percentage of time during which the high-mobility users stay on the macrocellular layer increases while the staying of low-mobility users are not affected. Without the setting of the Q_{offset} value (i.e. with Q_{offset} set to 0), the predominance of the microcellular layer is really evident: only 30% of high-mobility users is on the macro layer, and nearly 95% of low-mobility users is on the micro layer. With the lowest value of Q_{offset} , instead, the percentage of high-mobility users on the macro layer pass 65%. Moreover, in Figure 18 it has also to be noted that with Q_{offset} values lower than -10 dB the percentage does not increase more, as for Q_{offset} values higher than -4 dB the percentage does not decrease significantly. It could be concluded that in the simulated layout, the most affecting values of Q_{offset} are between -4 dB and -10 dB.

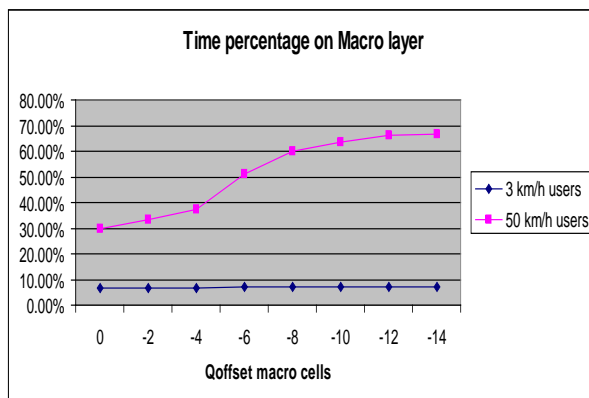


Figure 18.- Percentage of time of stay on the Macro layer – Micro with omni-directional antennas

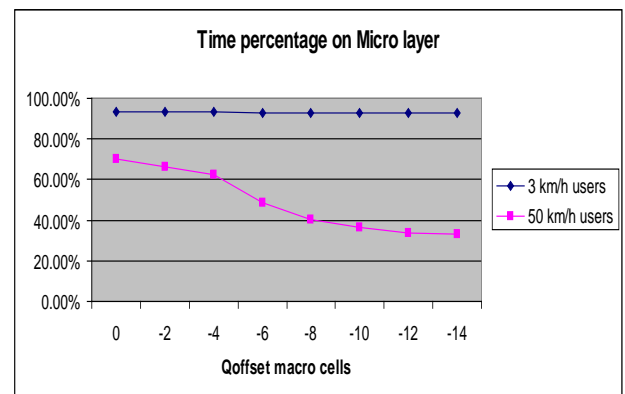


Figure 19.- Percentage of time of stay on the Micro layer – Micro with omni-directional antennas

Similar results are obtained when tri-sectorial antennas are considered.

In relation to the capacity analysis, Figure 20 shows traffic results, considering:

- Input load: the traffic offered to the system
- System load: the traffic carried by the system
- Blocked load: the blocked traffic (blocked when entering in the system)
- Dropped load: the dropped traffic (during the call).

It has to be noted that, decreasing the value of Q_{offset} , the blocked and dropped loads increase and then the system load decreases. The same phenomenon could be seen analyzing the blocking

² This specifies the offset between the serving cell and neighbor cell. It is used for FDD cells in case the quality measure for cell selection and re-selection is set to CPICH Ec/No.

percentage in case of new RAB setup, as depicted in Figure 21: both the uplink and the downlink blocking percentages increase with the increase of the segregation of the users on the layers according to their mobility class. Notice that, decreasing the value of Qoffset means to move users from the micro layer to the macro layer. Therefore all the high-speed users are moved from the micro to the macro layer. The users segregation leads to increase the “near-far” effect: in fact, considering a user camped on a macro cell, he could be closer to a micro cell antenna than a user camped on this micro cell, thus increasing the uplink interference. Furthermore, in downlink a similar situation is present: a user camped on a macro cell could be closer to a micro cell antenna than the antenna of its macro cell, thus increasing the downlink interference.

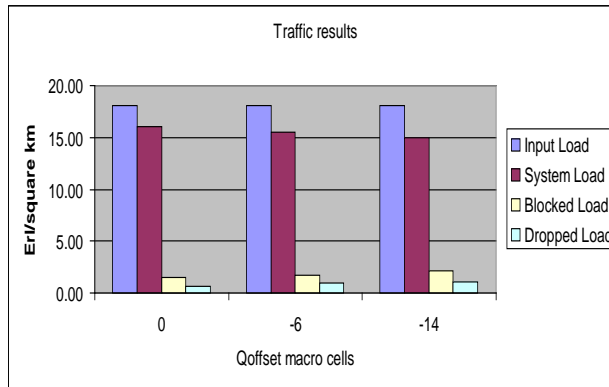


Figure 20.- Input load, System load, Blocked load and Dropped load results

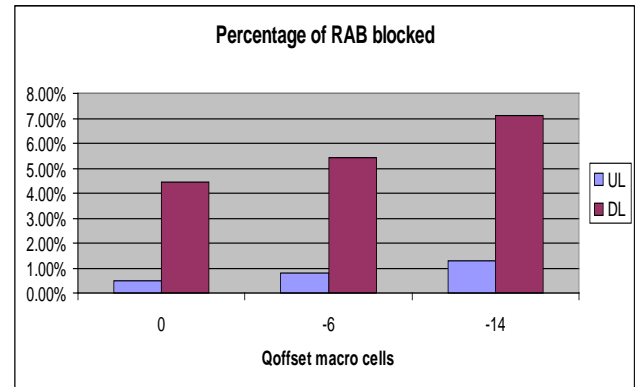


Figure 21.- RAB blocking percentages due to uplink and downlink congestion

The results of this analysis highlight that a network operator could manage the settings of a plethora of parameters in order to segregate the traffic according to its necessities. In the situation described in this study, the segregation has been based on the mobility class of the users. In addition, it has been also showed that an operator may set the trade-off between the negative effects of the segregation (e.g. impacts on the interference level due to the increase of the “near-far” effect) and the benefits of it, according to its objectives. The decision of which solution has to be used is left to be assessed case by case.

Another aspect that has been studied in HCS is the convenience of using or not different carrier frequencies for the macrocell and microcell layer depending on the traffic spatial distribution. In particular, let assume a scenario with different cells belonging to different HCS layers in which the spatial traffic distribution is non-homogeneous because of the existence of hot spots. Such hot spots may not in general be permanent but they will change in the different periods of the day and geographical locations, due to e.g. people concentrations at a bus-stop, restaurants areas during the meal times, etc. As a result, the load observed by the different cells as well as the interactions between them may experience significant variations during these periods. Although these issues may be covered at some extent by static radio network planning when devising the number and locations of the different cells and frequency allocations, the dynamics and randomness associated to the human behaviour may lead to situations in which the frequency allocation should be modified due to e.g. an excessive load in certain cells. Under this framework, a proposal based on the load gradient computations has been analysed to detect the cells having the highest influence over the rest of the cells in the scenario, thus being able to decide which cells should operate with a different carrier frequency, whenever a carrier frequency should be changed in the scenario. The proposed method has been compared with another scheme in which the cell with the highest load factor is selected to operate with another frequency. Results in terms of outage shown in Table 4 reveal for different case studies that the gradient methodology clearly outperforms the load-based algorithm as well as the case in which a single carrier is used in the scenario.

Table 4.- Outage probability in the different scenarios

	Equal Freq.	Gradient Alg.	Load. Alg.
Case 1	2.52 %	1.06 %	1.62 %
Case 2	28.92 %	18.28 %	27.99 %
Case 3	6.89 %	3.45 %	4.87 %

VIII- Transport Channel Type Switching

This section summarizes the work carried out with the aim to investigate the system-level performances of UTRAN when a Transport Channel Type Switching (TCTS) algorithm is taken into account. Notice that In the case of a discontinuous data transfer, it is very important to be able to optimize the usage of dedicated transport channel (DCH), preventing channelization code shortage in the downlink without degrading the end-to-end quality of service experienced by the user in an appreciable manner.

Figure 22 reports the total amount of traffic carried by the radio access network ("traffic load") in the two different cases, together with the corresponding amount of blocked traffic ("blocked load")³. The achieved results show that the UTRAN capacity is higher when the transport channel type switching algorithm is active. The capacity gain achieved by the switching algorithm can be justified taking into account the admission and congestion control procedures operated by UTRAN. With respect the admission control, the simulation results show that 3.8% of radio access bearer setup requests are blocked due to shortage of available downlink codes in the DCH-only case, whereas these events do not occur in the TCTS case, as represented in figure 23.

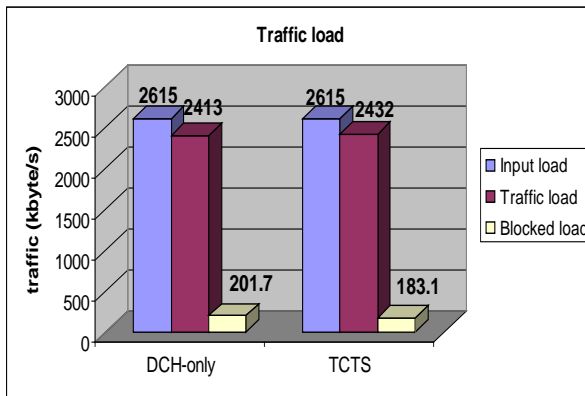


Figure 22.- Traffic results (DCH-only versus TCTS)

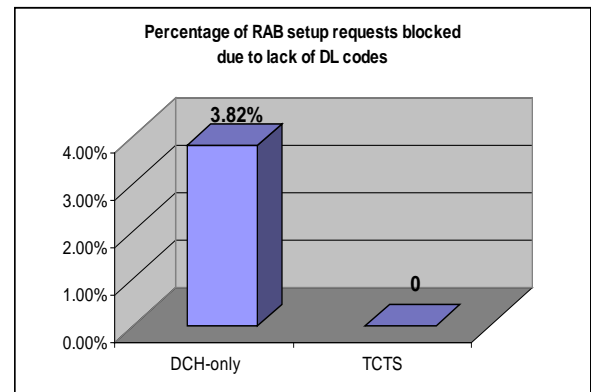


Figure 23.- Percentage of RAB setup requests blocked due to lack of DL codes (DCH-only versus TCTS)

The difference in the achieved results between the two cases can be understood taking into account that in DCH-only mode, each DCH has its own specific OSVF code corresponding to an appropriate spreading factor assigned for the whole duration of the connection. If the number of users requiring DCHs becomes high, the code shortage cause the block of some users.

In summary, and as a general rule, even if the common channels are used only when a very small amount of data (or nothing at all) have to be transferred, the QoS is better for DCH-only than for TCTS because of the switching time. DCH offers higher transfer speeds (throughput) but requires a significant setup time, whereas shared channels have a low throughput but also a low setup time. Due to these considerations the usage of common channels is more efficient only when the traffic is sporadic or for short time, otherwise for long and frequent data sessions, the usage of a dedicated channel is highly recommended. Another very important issue consists in the proper setting of the parameters controlling the reporting of the traffic measurements performed by the users, in order to prevent an overhead of useless signalling traffic.

IX- DiffServ Aware Scheduling

This section focuses on the colour aware RRM algorithms to support DiffServ over CDMA radio systems. The schemes are designed to use the IP layer traffic information to increase the utilization of radio network. The critical impetus of the proposed approach, beyond minimization of the transmitted power, is mainly twofold. Firstly, to achieve the required per-class aggregate data rate while prioritizing and ensuring QoS of in-profile packets, and secondly to increase power gains by penalizing out-of-profile packets in sense of power consumption. The seminal aspect of the proposed scheme is that tangible power gains can be achieved by differentiating transmission of conformant and non-conformant

³ Taking into account that the users have been distributed uniformly in all the 36 cells of the simulated scenarios, the mean amount of traffic in input per each cell is approximately equal to 600 kbit/s in both cases, as reported in 3.4.

packets while at the same time the aggregate power gains of AF classes can be utilized to enhance the performance of in-profile traffic.

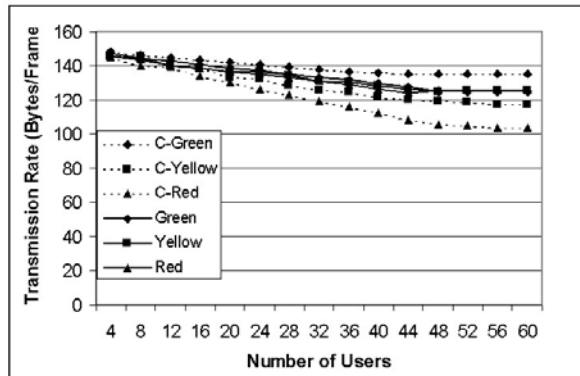


Figure 24.- Transmission Rate.

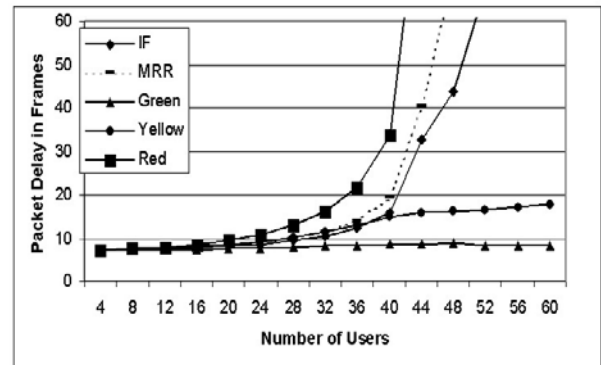


Figure 25.- Delay Performance of IF and MRR with color aware.

Results in Figure 24 clearly show the differentiation in resource allocation in terms of transmission rate with the proposed scheme. In this figure as expected, the transmission rates decrease for all type of users as traffic load increases. And the transmission rates for all types of packet are almost the same without color aware shown by the solid line. However in the case with color aware queue, the transmission rate for each color is different to each other. Green users always have a better rate than other users because they are given high priority than other users. Also as traffic load increases, the difference becomes larger. The red users' transmission rate decreases more rapidly (from more than 145bytes drops to just over 100bytes) and in the contrast, the green users rate are only drop about 10bytes. Figure 25 shows the delay performance of MRR (Modified Round Robin) and IF (Interference Factor)⁴ queue both with color aware. IF offers a better performance in the average delay. The diversity of delay performance for different colored packets shows the consistence with the differentiation in transmission rate allocation presented in Figure 24. Again to enhance the delay performance of the in profile packets such as green packets, the out profile packets such as red packets are penalized in high traffic load situation.

X- High Speed Downlink Packet Access(HSDPA)

The main benefit with the high-speed downlink packet access (HSDPA) evolution of the WCDMA is that it reduces the user packet call delay and it increases the system capacity in downlink. This increased capacity can be used to either increase the number of users in each cell, and/or to provide the existing users with higher average data rates. It is important to note that the trade-off between HSDPA system throughput versus QoS is critical, and several different parameters needs to be taken into account when designing RRM strategies for HSDPA. The scheduling method used to decide which user is allocated to the channels is of great importance. The HSDPA scheduling is very flexible. It allows sharing of the spreading codes (here denoted as "channels"). This means that one or several channels are allocated to one user and several users may receive data under one transmission time interval (TTI). It is also possible to schedule all channels to one user under one TTI, and this is considered as the primarily means of sharing.

Advanced schedulers, that combine both channel information and service requirements, have been evaluated and the results are presented in terms of satisfied users and cell throughput both for indoor and vehicular environments. In particular, scheduler algorithms were proposed for the video streaming and are based on weighting parameters that affect the application functionality and the system (cell) throughput. Complementary to the scheduler, the RRM allocation policy tries to maintain flexibility by assigning several users to a single TTI, scheduling individual packets.

⁴ At each frame, the users are ordered according to their propagation conditions. The user with the best propagation condition will be placed on the top the queue

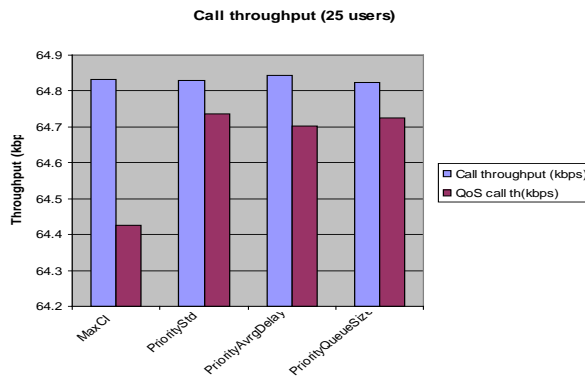


Figure 26. User average bit rate – Comparing all schedulers

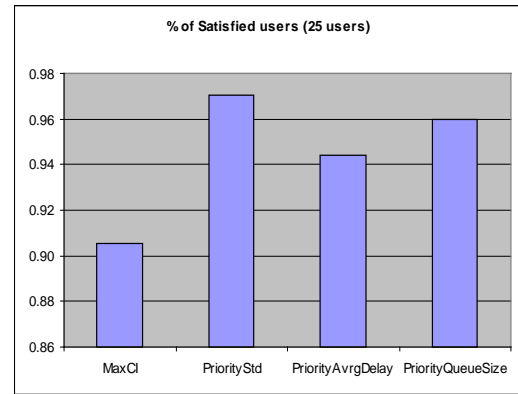


Figure 27. Percentage of satisfied users – Comparing all schedulers

Comparing the schedulers, concerning QoS, better results both for QoS bit rate and satisfied users, were obtained with the Priority Standard scheduler, P_1 . The Max. CI scheduler is the one that lead to worst QoS results, since it does not consider services constrains.

In addition to that, a feasibility study on supporting VoIP over HSDPA by using biased AMC scheme was also carried out. Dynamic simulation results showed that a system capacity of 70 Erlangs can be achieved at velocities up to 20 km/h, by combining a longest delay first (LDF) discipline with a proportionally fair (PF) code multiplexing scheduler, and by using a block size of 20 to 80 ms. However, at 80 km/h the capacity is reduced to 12 Erlangs due to the inability to track fast fading with CQI feedback delay; the AMR and scheduler has to rely on outdated CQI reports. Observing that the channel estimate is less accurate when the channel SIR is low, a biased AMC scheme was further introduced. The obtained simulation results showed that adapting the offset by the average SIR (representing location) produces a marginal significance, and a fixed bias is sufficient to improve the VoIP QoS. This implies that the CQI feedback delay is the main evil that must be mitigated. The results also showed that the biased AMC increases the system capacity by over a three-fold at 80 km/h.

Moreover, HSDPA is capable of supporting various QoS levels. In fact, the HSDPA specification supports up to 16 priority levels and 8 queues per user. Then, when heterogeneous traffic coexists, an innovative HSDPA packet scheduler that discriminates QoS classes was proposed to provide better QoS support than those that consider only the channel condition. Three services are considered: voice over IP (VoIP), variable bit rate (VBR) video streaming (MPEG, hereafter), and web-browsing (HTTP, hereafter). The proposed scheduler was shown to support the heterogeneous traffic efficiently, without causing any particular service being a bottleneck in providing capacity. The simulation results further showed that the proposed scheduler is robust to changes in the traffic constitution, and supports call arrival rates of up to 3.5 calls/s in guaranteeing 5% outage to all the services in the default traffic scenario. This is substantially larger than some conventional schedulers, as strict priority queuing schemes support only up to 2.8 calls/s, whereas single queue schemes fail to secure 1 call/s, according to the simulation results.

XI- RAN Sharing

Sharing spectrum can be very attractive, for example, in rural areas UMTS coverage can be offered with much lower investment costs, but also in urban areas and hot spot areas capacity gains can be achieved. Certainly, there is a capacity gain due to the increased trunking efficiency as channels are pooled together between the operators. However, Fairness of resource allocation can become an issue when operators share spectrum.

Table 5. 50% of users from Operator A and 50% from Operator B

	No admission control	Total power with dropping	Half power	Multi-cell
Admission (%)	100	91.20	83.89	89.35
Dropping (%)	6.63	1.53	0.16	0.67
GoS	66.39	24.08	17.7	17.39 [G=1.75%]

Table 5 shows the admission and dropping percentages when 50% of the users are from Operator A and 50% from operator B for the different algorithms. Moreover, the Grade of Service⁵ is also presented. As shown in Table 5, the *Multi-cell*⁶ algorithm provides the best Grade of Service, because it provides higher admission probability than the *Half power*⁷ algorithm and maintains the dropping probability in a low value. As shown in brackets the gain in Grade of Service of the *Multi-cell* algorithm with respect to the *Half power* is $G=1.75\%$. Moreover, it can be observed that the *Total power with dropping*⁸ algorithm provides a poor Grade of Service because the droppings forced by this algorithm increase the dropping probability.

XII- Location Aware Resource Reservation

Recent developments in the positioning technology in the context of WCDMA systems provide strong assurance that accurate position measurements will become a viable reality in the near future. In that context, the location information obtained by these techniques may provide better predictions of future resource availability, and therefore, it can be exploited to develop more advanced RRM (Radio Resource Management) strategies that increase the system efficiency. Certainly, the knowledge of the location and mobility pattern of the users will provide certain information to estimate the future need and availability of the radio resources. In particular, in scenarios with users moving along a road, these location estimations can be more accurate because main road users have usually a straight mobility pattern. Therefore, more accurate predictions of handover requests can be done, and consequently, certain radio resource mechanisms can be triggered in order to assure available resources for the handover procedure, increasing the efficiency in the use of the system resources.

The main objective of the proposed reservation algorithm is to assure service to business users moving along a main road while at the same keeping the service of consumer users at a satisfactory level. To this end, the proposed algorithm defines a certain reservation region around each station, starting at the reservation distance D . As shown in Figure 28, there is an optimum value for the reservation distance that minimises the system GoS. It can be observed that a reservation distance between 1300 and 1700 metres provides the lower GoS values. It is worth noting that a too high value of reservation distance may cause that a main road business user with reservation for a given cell may end its current connection before the handover is eventually made effective (i.e. false reservation). In this situation, the reserved resources for this user have been wasted, reducing the system efficiency. The false reservation probability is shown in Figure 29. Higher reservation distance causes a higher false reservation probability, particularly for short call durations.

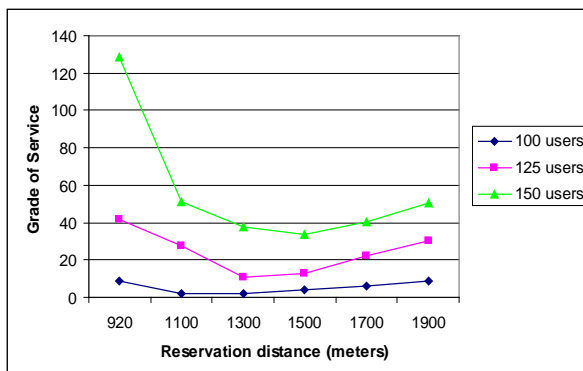


Figure 28. Grade of Service for different reservation distance (mean call duration 120seconds).

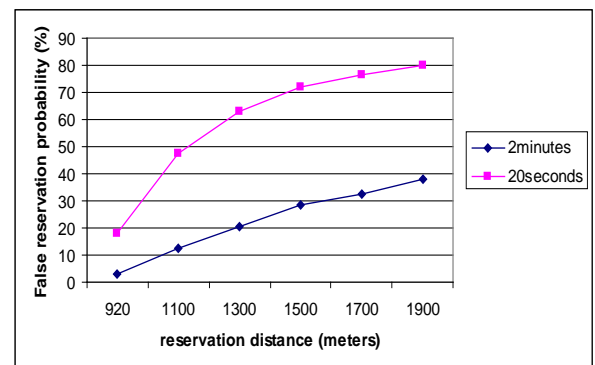


Figure 29.- False reservation probability for different reservation distance.

⁵ The Grade of Service is defined as $GoS = Pb + 10Pd$ where Pb is the blocking probability and Pd is the dropping probability

⁶ The multi-cell algorithm not only considers the power availability of a single base station but the power availability of neighbouring cells when an admission request is processed.

⁷ When an operator A user makes an admission request, it is accepted if the total power devoted to this operator is lower than half of the maximum power

⁸ When an operator A user makes an admission request, it is accepted if the total power devoted to this operator is lower than the maximum power. In the opposite case, if operator B is using more than half of the total power, the connection request is admitted and certain number of Operator B users must be dropped in order to make room for the new Operator A user.

The proposed algorithm optimizes the existing trade-off between the dropping probability and the blocking probability in a WCDMA system. The reservation of certain resources for handover users reduces the number of dropped connections at expense of certain increase in the blocking probability of new connection requests. The proposed algorithm takes advantage of the predictability in the movement of users along a main road in order to determine the most adequate instant of time when the resource reservation for handover users should be made. A too large reservation region may cause that a handover user ends its connection before starting the handover procedure, resulting in a high false reservation ratio. Moreover, the blocking probability of new connection requests would increase because a large number of resources would be devoted to reservations for users inside the reservation region. On the other hand, a too small reservation region increases the number of handover failures (i.e. the dropping ratio) because there is not time enough to obtain the available resources. Then, an optimization of the reservation distance has been made by minimizing the system GoS. Moreover, the impact of the user call duration and service bit rate on the proposed algorithm has been discussed. It has been shown that, for shorter call durations, the reservation distance must be lower in order to reduce the false reservation probability. Similarly, higher service bit rate require higher reservation distance because more time is needed in order to obtain the required resources.

RRM issues for GERAN

The most important radio resource management mechanisms offered by GPRS/EGPRS to support video streaming services have been identified and investigated, analyzing the main procedures related to QoS handling and channel administration for the services belonging to the streaming class, as well as all the standard features and parameters of the system able to affect the quality of these types of services.

The aim of the carried out simulation analysis was to set the value of all the most important parameters which are not imposed by the GPRS specifications, in order to optimize the performances both of the network and the streaming clients. In particular, one important degree of freedom, which has been tuned in our simulation experiments, is the RTP packet size, which is the amount of segmentation introduced by RTP protocol. As a matter of fact if we use RTP PDU packets which are too long some disadvantages have to be taken into account:

- The latency time needed for receiving a single packet gets longer
- When a long RTP packet is lost, a lot of information is lost or must be transmitted again
- The jitter due to delay variation is higher if a retransmission scheme is adopted
- To compensate the previous effects a longer buffer must be used in the client

Figure 30 shows the simulation results which are obtained when the application throughput is plotted as a function of the RTP packet size. As we can see the application throughput is nearly constant and equal to its maximum value when the dimension of RTP packet is greater than 400 bytes. If we reduce this dimension up to 100 bytes the throughput decreases from its maximum value and the effects of a too fine segmentation in packets are too strong. This is of course more evident when the codec bit rate is higher, whereas for smaller dimension of RTP packets the dimension of the packet may be still reduced if the overhead is still acceptable.

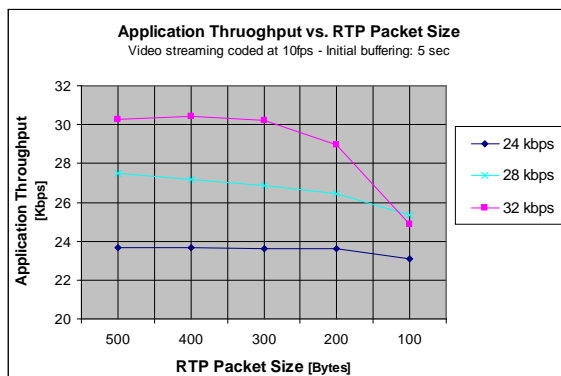


Figure 30.- Application throughput versus RTP packet size

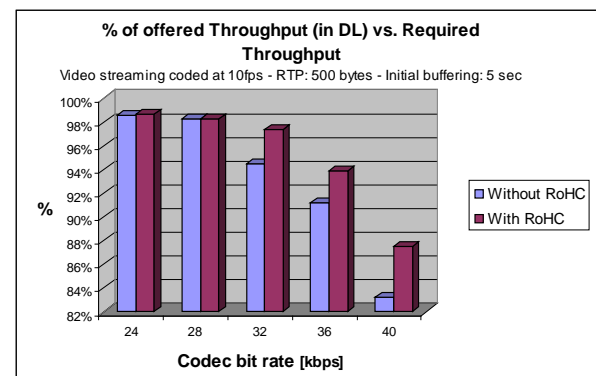


Figure 31.- Offered Throughput in DL vs. Required Throughput with/without RoHC option

It should be noted that there is an advantage in limiting the maximum size of RTP packets to avoid SNDCP fragmentation over the GERAN Gb interface where the size for LLC data file cannot be too large (default size is 500 bytes). On the other hand, when RTP packets are chosen to be too small, the packets could result in too much header overhead. In order to reduce the effects of the considerable header overhead for voice and video applications the optional functionality for compression of RTP/UDP/IP headers is also supported. In particular, the header compression scheme applied is the called RObust Header Compression RoHC, which allows reducing the RTP/UDP/IP header size from 40 bytes to 3 bytes.

From another point of view, figure 31 plots the offered throughput in DL considering both cases, with and without RoHC. As here depicted, for requested low bit rates (24 and 28 kbps) the system is nearly able to offer them (approximately 98.55 %), so the contribution of header compression is lower. Whereas, if we look from the network capacity viewpoint, for high bit rates the compression of excess protocol headers leads to an increase of the offered throughput (for requested 32 kbps the offered throughput increases from 94.53 % without RoHC up to 97.40 % with RoHC), so that, this is still growing more closer to requested throughput than experiments without RoHC.

Finally, the effect of the initial buffering time on the quality perceived by the user is analyzed. Taking into account that codec bit rates from 24 up to 32 kbps have been defined as the acceptable bit rates for video streaming applications over GRPS networks, in figure 32 two parameters are measured, such as the

average number of blocks per session and the percentage of satisfied users, all produced when the initial buffering time is varying from 1 to 5 seconds.

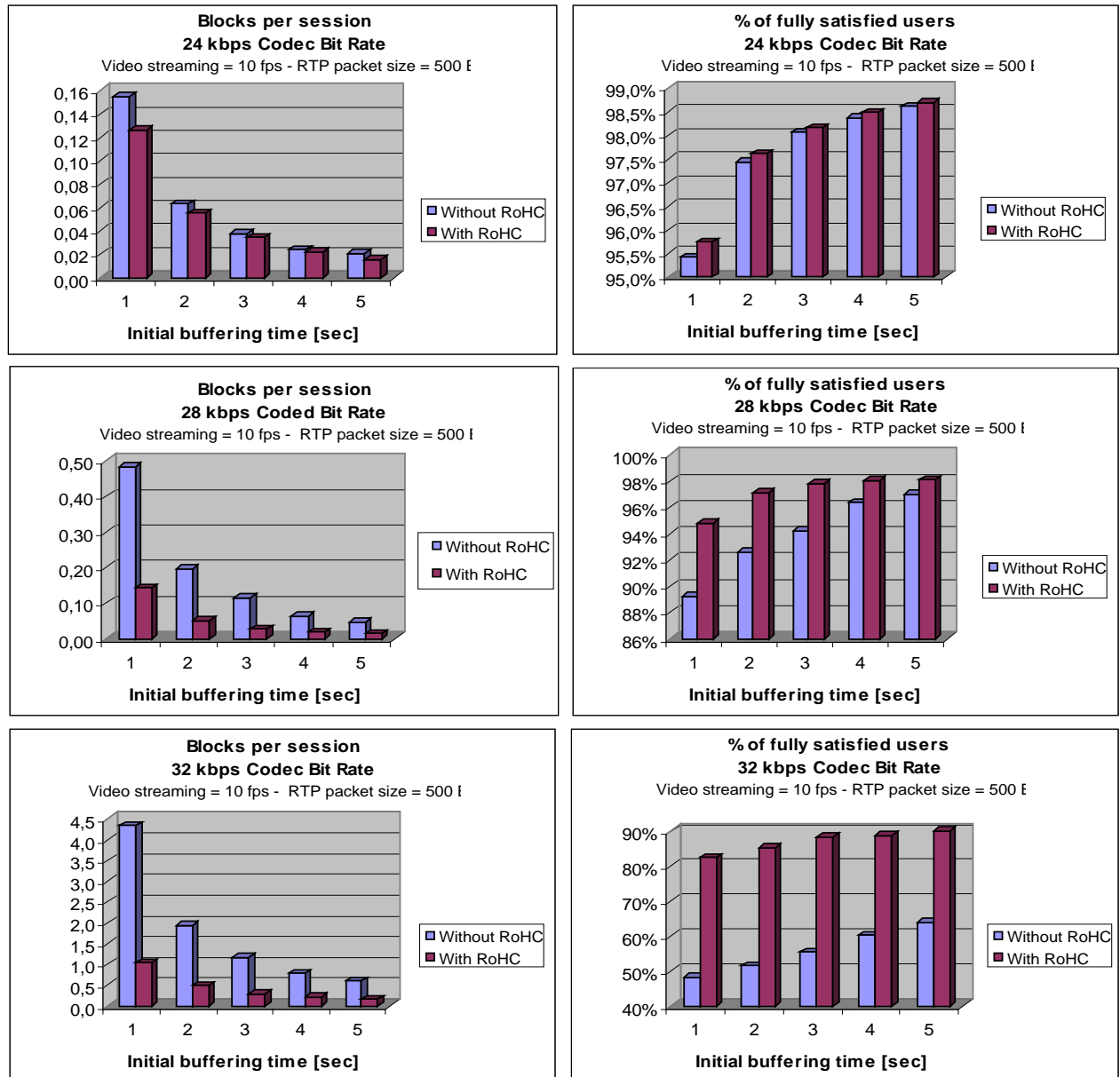


Figure 32.- Average number of blocks per session and percentage of fully satisfied users vs. initial buffering time (24, 28 and 32 kbps codec bit rates)

From the figure 32 we can notice that as buffering time is increasing, the blocks per session are decreasing, and consequently the percentage of fully satisfied users is increasing. Furthermore, as we have seen the results are enhanced when it is applied the header compression mechanism. Hence, taking as reference the case with RoHC option, we are able to establish a minimum initial buffering time, which compensates the jitter effects and maintain a good level of user satisfaction. In fact, a safe value for the initial buffering time must be 5 seconds.

In summary, the simulation results show that low-loaded GPRS cells are able to support video streaming with an offered throughput from 24 kbps to 30 kbps, even in presence of GSM users (if sufficient amount of radio resource are available in the cell and these ones are managed in the most appropriate way). On the contrary, when the total amount of radio resource offered by the cell is not enough to support all the active users (GSM and GPRS), the level of decrease in the QoS of the streaming sessions has been measured. Other simulation results stress the importance to consider some specific mechanisms able to maximize the quality perceived by the video streaming end-user, like client-side pre-bufferization and header compression techniques.

RRM issues for WLAN

This is particularly true for the Wireless LANs of IEEE 802.11 family, in which the way of work of the Medium Access Control (MAC) layer, according to the so called Distributed Coordination Function (DCF) access method, make the system's performances very sensitive respect to the number of stations and their traffic profiles. On the basis of these considerations, it should be evident that without any policy for blocking the admission of new users when specific load conditions occur, wireless LAN systems cannot succeed to guarantee whatever profile of Quality of Service and only best effort services can be supported. In this framework, the WLAN-related issues dealt in EVEREST so far and reported below correspond to:

- An analytical model for an admission control algorithm, accompanied by results on admission control regions for a mix of services
- A short to medium term proposal for QoS enhancements on 802.11b based on the Hierarchical Token Bucket algorithm
- A more long term evaluation for QoS enhancements on 802.11e

I- Admission Control for IEEE 802.11A/B/G

The proposed Admission Control (AC) policy consists in three main steps:

- Step 1: throughput offered by the BSS to each real-time user considered by the EVEREST project is evaluated by means of an analytical model [5]
- Step 2: results coming from the analytical model are exploited in order to identify the maximum number of users for each class (i.e. services mix) that the considered WLAN BSS can support ("Capacity region"; see figure 33);
- Step 3: the algorithm in charge of the AC policy can be based on the capacity region related to the WLAN BSS. More in detail:
 - For every stations requesting a new association to the wireless LAN Access Point, retrieve the type of service requested by the user;
 - If the acceptance of the new user brings the load work point of the WLAN BSS outside the QoS region (according to the a-priori evaluated capacity region), deny the request of association;
 - If the new user can be admitted without compromising the minimum level of offered throughput per each class of users, accept the request of association and update the load work point of the system (i.e. number of active users within the systems).
 - For every stations requesting a dissociation from the wireless LAN Access Point, update the load work point of the system (i.e. number of active users within the systems);

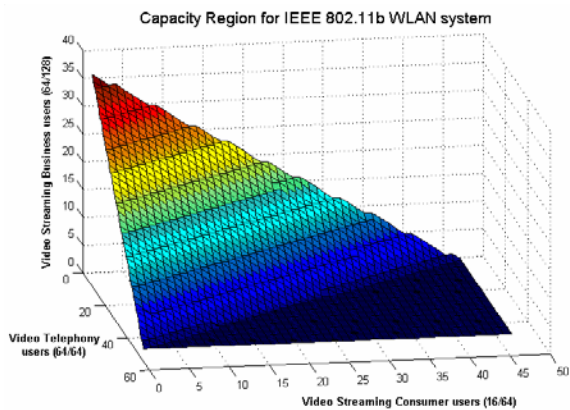


Figure 33.- Capacity region offered by an 802.11b hot-spot for the real-time services envisaged by EVEREST scenario.

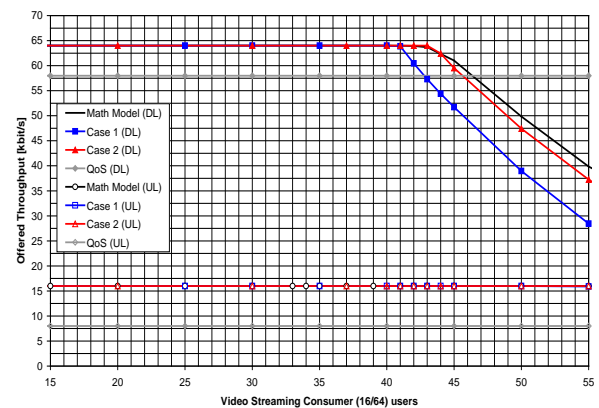


Figure 34.- Offered throughput per user (UL/DL) versus number of Video Streaming Consumer users

The comparison, in terms of offered throughput, between the simulation results and those calculated from the analytical model are shown in Figure 34. Notice that, the small differences perceived between the two are related to some approximations of the analytical model (for example the Acktimeout and the beacon).

II- Admission Control for IEEE 802.11E

Concerning the admission control algorithms for wireless local area network (WLAN) IEEE 802.11e Enhanced Distributed Channel Access (EDCA), a new admission control algorithm called Enhanced Distributed Admission Control algorithm (EDAC) was proposed. The algorithm is able to protect already active flows of continuous nature (conversation and streaming) and allows controlling low priority bursty traffic by means of minimum average guaranteed load. It provides a dynamic control of time spend on transmissions from each access category managed by Transmission Time Threshold parameter and controlled by AP.

In that sense, the aggregated throughput for voice traffics with and without admission control was analysed and the results are shown in Figure 35. We observe that without the EDAC algorithm the throughput starts to oscillate when system load is very heavy (18 stations). The proposed algorithm stops further admission of voice stations and prevents the system from entering into the saturation state. Certainly, without admission control all stations are allowed to enter the system. Therefore when the system becomes saturated (overloaded) the collisions between packets increase and as a result, the aggregated throughput for voice traffic fluctuates. In consequence, in the saturation case, without EDAC algorithm the QoS requirements for voice traffic cannot be guaranteed. On the other hand, with admission control mechanism, when the system reaches the heavy load state no more stations are admitted and already active stations are protected.

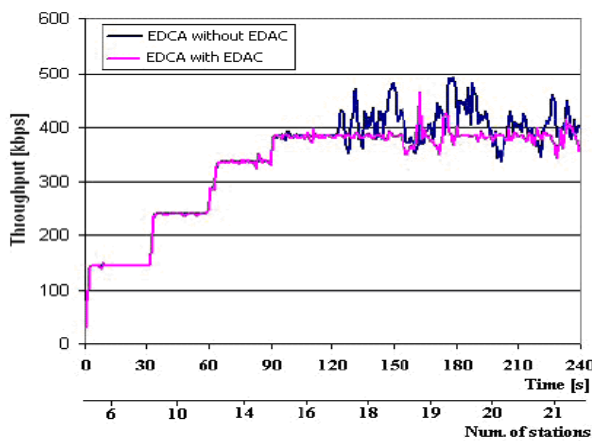


Figure 35 - Aggregated throughput for voice traffic with and without EDAC algorithm

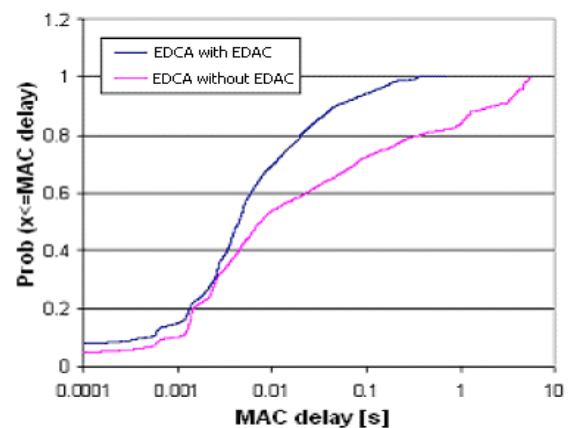


Figure 36- CDF for video MAC delay with and without EDAC algorithm

Next, Figure 36 demonstrates the cumulative distribution function of MAC delay for video traffic with and without admission control. Comparing these two plots we may clearly see that without admission control the MAC delay of 95% of packets is lower than 4s, whereas in case of EDAC mechanism MAC delay for the same case is lower than 0.11 s. Accordingly, uncontrolled admission of video stations in the set-up scenario without EDAC algorithm provoke a significant increase of experienced delay of these stations and, in consequence, the loss of their QoS expectative.

III- On the use of the EDCA Transmission Opportunity (TXOP) mechanism for improving the WLAN system performances

In the standard IEEE 802.11 as a result of varying radio channel properties the station changes its transmission bit rate at the physical layer to increase its resilience to experienced errors. However, when changing physical layer it also changes the bandwidth required for transmitting a packet, which, in case of downward shifting of physical layer, may provoke a congestion problem in the system as required bandwidth increases. The new draft "e" of the standard IEEE 802.11 provides a novel mechanism for packet transmission allowing multiple packet transmission by the station once the channel has been captured. This mechanism is called Transmission Opportunity (TXOP) and it is characterised by its start time and duration. The new draft "e" of the standard IEEE 802.11 provides a novel mechanism for packet transmission allowing multiple packet transmission by the station once the channel has been captured. This mechanism is called Transmission Opportunity (TXOP) and it is characterised by its start time and duration. This enhancement may significantly improve the system performance as it optimizes the channel efficiency by allowing successive packet delivery. Moreover, it solves the problem related to the unknown transmission duration of the legacy IEEE 802.11 stations due to changes in the bit rate value (link adaptation mechanism). Furthermore, by applying different TXOP duration times to the different traffic types, some QoS control over these flows may be obtained.

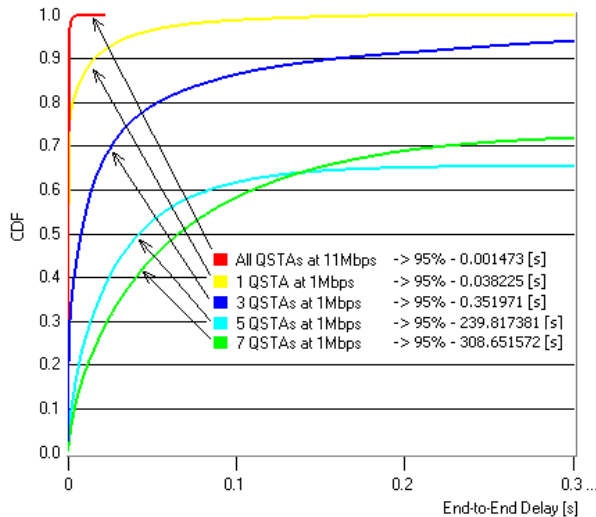


Figure 37.- Cumulative distribution function of end-to-end delay at link layer level of aggregated voice stream. Without TXOP

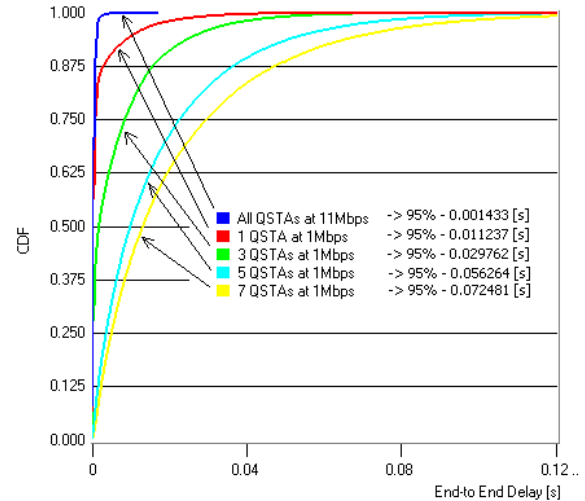


Figure 38.- Cumulative distribution function of end-to-end delay at link layer level of the aggregated voice streams. With TXOP

Comparing the accumulative distribution function of the delay of the of aggregated voice streams with (Figure 38) and without (Figure 37) TXOP mechanism it could be realized that the TXOP mechanism has significantly enhanced the system performance. Certainly, Figure 38 shows how now, with a TXOP limit, the experienced link layer delay by voice streams is lower than the obtained without TXOP limit (see Figure 37). For instance in the case of $3@1^9$ the obtained bound delay for the 95% of the cases is 0.0298 sec. with TXOP whereas without TXOP this limit reaches 0.3520 sec.). This improvement is due to the fact that when using packet bursting there is lower number of packets contending for a channel access. Moreover, also notice that the obtained delay values for voice traffic are within required limits (less than 100 msec.) even in deep saturation scenario $7@1^{10}$ when TXOP mechanism is used.

Moreover, simulation results show that there exist optimum TXOP limit values in terms of system performance. However, as these values change with service mix distribution a novel dynamic TXOP configuration algorithm based on the number of packets in AP queues has been proposed. The developed algorithm provides TXOP limits that are within minimum most advantageous TXOP limits and hence quasi-optimal system performance is reached independently of system service mix.

Finally, by using the TXOP limit to perform fragmentation of low priority packets, further enhancement of prioritization mechanism is obtain.

Evaluation of CRRM algorithms

Common Radio Resource Management (CRRM) refers to the set of functions that are devoted to ensure an efficient use of the available radio resources in heterogeneous networks scenarios by means of a proper coordination between the different radio access networks.

With respect to Common Radio Resource Management, a first contribution of the project has been the outline of a functional model for having a common management of the pool of radio resources in heterogeneous scenarios. In the presented model, the Common Radio Resource Management (CRRM) refers to the set of functions that are devoted to ensure a proper coordination between the different radio access networks to achieve the most efficient use of the available radio resources. Different approaches to the CRRM and RRM interaction have been presented, outlining the potential levels of coordination in the radio resource management decisions in the identified functionalities. Finally, the requirements in terms of interworking capabilities and considerations about the physical CRRM implementation have been detailed. According to this framework, and taking into account the scope and time-frame of the EVEREST project, the CRRM studies reported here assume the functionality split in which the CRRM takes charge of the RAT selection procedures, including both initial RAT selection and vertical handover, while the RAT-specific RRM algorithms, like admission control, congestion control or packet scheduling are executed locally at the RRM entities.

⁹ $3@1$, 3 web stations transmit at 1 Mbps and the rest at 11Mbps

¹⁰ $7@1$, 3 web, 2 video and 2 voice stations transmit at 1 Mbps. and the rest at 11 Mbps;

I- Service-Based RAT Selection Policies

A general policy-based framework for the specification of CRRM algorithms has been defined and different policies considering the service type as well as the fact that the users may be indoor or outdoor have been evaluated through simulations. In order to stress the influence of the initial RAT selection procedure, without taking into consideration handover issues, no vertical handover strategy is initially considered in these simulations, Table 6 presents the aggregate throughput (in Mbit/s) achieved with the sum of both RATs (GERAN and UTRAN) and with the sum of both services (voice and www) for the two considered basic service-based policies: Voice UTRAN (VU) and Voice GERAN (VG).

Table 6.- Aggregate throughput for the different policies

	VU				VG				VG (no TrCH switch)			
	UL		DL		UL		DL		UL		DL	
www users	0.5 Km	1 Km	0.5 Km	1 Km	0.5 Km	1 Km	0.5 Km	1 Km	0.5 Km	1 Km	0.5 Km	1 Km
200	2.18	2.08	2.22	2.17	2.14	2.14	2.20	2.22	2.03	2.01	2.08	2.07
600	3.01	2.88	3.15	3.09	2.96	2.95	3.16	3.15	2.06	2.05	2.11	2.11
1000	3.80	3.64	4.05	3.96	3.77	3.76	4.08	4.08	2.08	2.05	2.14	2.13

It has been obtained that, in outdoor scenarios, VG basic policy turns into a higher throughput than VU, ensuring lower interactive packet delay. This is because of the higher efficiency for non-real time traffic transmission in UTRAN achieved in the VG case, since web browsing traffic is supported by means of dedicated channels whereas in VU a packet scheduling algorithm must be implemented in GERAN.

Another type of basic initial RAT selection policies are those that take into account radio network considerations of each specific RAT. For instance, WCDMA capacity is highly degraded by indoor traffic users, which leads to defining a new basic policy, denoted as IN (indoor) in which indoor users are allocated in GERAN and outdoor users in UTRAN. By combining the basic policies VG,VU and IN, the following complex policies have also been studied:

- VG*IN→ If the service is voice and the user is outdoor, the first choice will be to allocate it in GERAN (i.e. according to the VG policy). If no capacity is available in GERAN, the second choice will be to allocate it in UTRAN. If the service is www and the user is outdoor, a blocking will occur if there is not capacity in UTRAN.
- IN*VG→ In this case, the first choice takes into account whether the user is indoor or outdoor and, if no capacity is available in the selected RAT, the second choice considers the service type.
- VG*VU→ According to this policy, voice users will first fill the capacity available in GERAN and then they will be directed to UTRAN. In turn, www users will first fill the capacity in UTRAN and then they will be directed to GERAN. In this case, no request is blocked provided that there is capacity available in either UTRAN or GERAN

In scenarios with a mix of indoor and outdoor users with different services, up to medium voice loads (i.e. 200 users) no relevant differences between the policies are observed, although in general the performance of IN*VG is somewhat poorer, mainly when the number of www users increases. The reason is that, with IN*VG, there is a higher number of www users that are served through GERAN (i.e. those that are indoor), which provides higher delays and lower www throughput than UTRAN (see Figure 39 and Figure 40). Further, when the ratio of indoor users increases, the number of interactive users allocated in GERAN also increases and, consequently, IN*VG performance is more degraded.

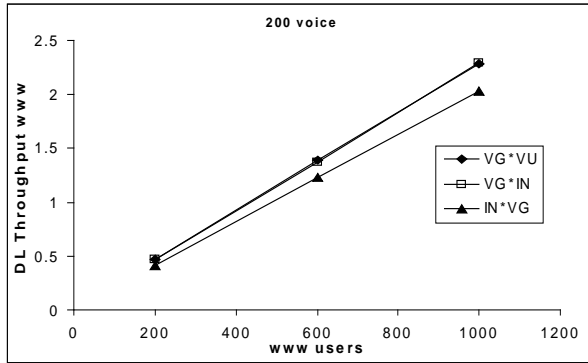


Figure 39- Total DL www throughput with 50% indoor traffic

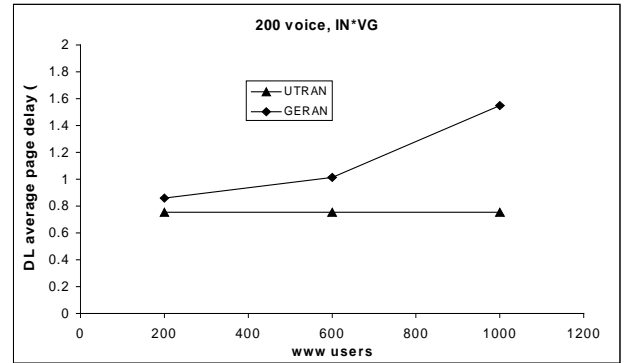


Figure 40- DL average page delay with IN*VG policy and 50% indoor traffic

In summary, the performance of IN*VG policy improves when the voice load increases, the www load decreases and there is a high fraction of indoor users. On the contrary, for low voice loads and high www loads VG*IN achieves a better throughput. This suggests that the suitable configuration of the RRM and CRRM entities according to specific policies depends on the existing traffic conditions and therefore it may be modified at e.g. different periods of the day.

With respect to vertical handover, the interworking between horizontal and vertical handovers has been studied, with two considered approaches, namely the tight approach T-VHO, in which the vertical handover algorithm is executed at every time that a horizontal handover algorithm should be carried out, so that both possibilities are considered prior to taking a decision, and the loose approach L-VHO, in which the vertical handover algorithm is executed only when a horizontal handover fails or when a call dropping is about to occur due to bad propagation conditions.

When comparing T-VHO with L-VHO, the first aspect to consider is how the traffic distribution changes in the different RATs. In particular, when 400 voice users are present in the scenario, Figure 41 shows the percentage of voice traffic served through UTRAN as a function of the number of www users. In this case, even when no vertical handover is available, a certain portion of the voice traffic is served through UTRAN, corresponding to the new voice calls that are originated in a blocked GERAN cell (for this voice traffic load, the probability that this occurs is found to be around 0.5%). In turn, when vertical handover is used, a certain number of voice users are also transferred from GERAN to UTRAN when they reach blocked cells during horizontal handovers, thus increasing the fraction of voice traffic served through UTRAN. Notice that with the tight approach there are less voice users in UTRAN than with the loose approach, because in the latter users that are handed over to UTRAN will tend to remain there, while with the tight approach a voice user in UTRAN will try to return to GERAN in the first UTRAN horizontal handover.

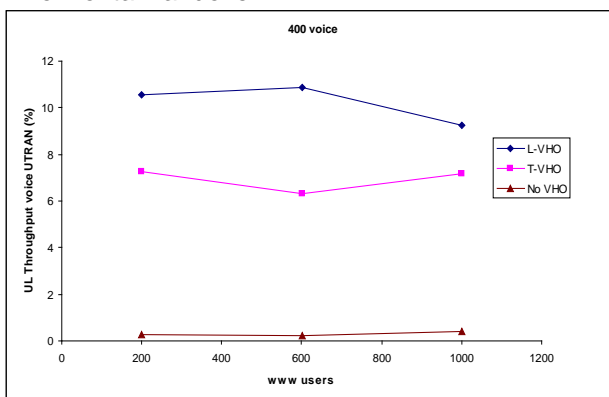


Figure 41 Percentage of voice throughput served through UTRAN for the 400 voice users case

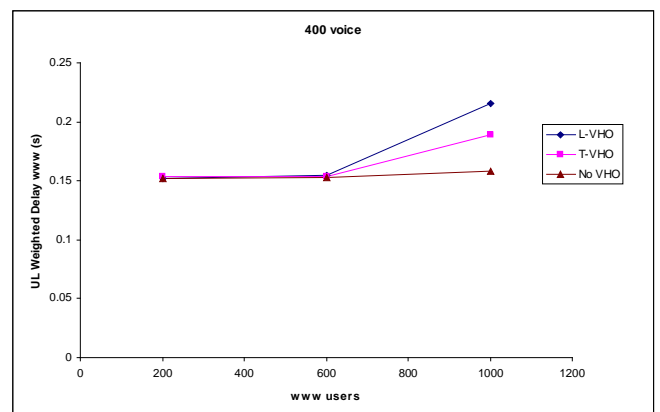


Figure 42 Uplink weighted delay for www traffic with 400 voice users

Notice that the tight vertical handover approach facilitates the fulfilment of the service policy, serving the www traffic as much as possible through UTRAN. This has an important impact in terms of weighted average delay for www traffic, as reflected in Figure 42

In summary, it has been shown that the traffic distribution among the considered RATs can be quite different with the two approaches. In that sense, the tight approach allows a better fulfilment of the initial

RAT selection policy, due to having more chances to execute a vertical handover than in the loose case. Consequently, and for a service-based RAT selection policy, it has been observed that the tight approach offers a better performance in terms of lower delay for interactive www users than the loose approach because most of the www traffic is served through UTRAN. On the other hand, a higher number of vertical handover procedures are also required with the tight approach, which increases the signalling overhead.

II.- Load Balancing -Based RAT Selection Policies

Load balancing (LB) is a possible guiding principle for resource allocation in which the RAT selection policy will distribute the load among all resources as evenly as possible. Taking this into account, the performance of load balancing principles in the RAT selection procedure has been covered and compared against a service-based policy.

Figure 43 shows the average cell load of the central base station in both RATs for SM1¹¹ when policies VGVU and LB are used. Note that for VGVU policy, voice users are directed to GERAN while not fully-loaded; otherwise, requests are transferred to UTRAN. Load balancing policy LB behaves as expected, maintaining cell load levels in both RATs at approximately the same level. Figure 44 shows the load levels for policies VGVU and LB for SM2. For this service mixing, policy LB does not show a visible improvement with respect to the service class policy VGVU. In fact, the load levels are kept low, compared to SM1, and therefore no severe dropping occurs.

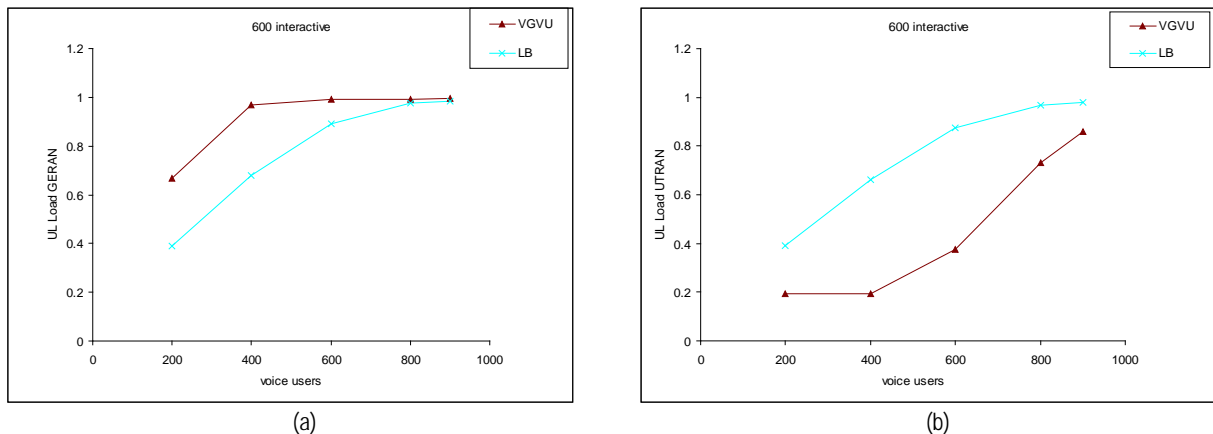


Figure 43- Average UL load in (a) GERAN and (b) UTRAN for policies VGVU and LB with SM1

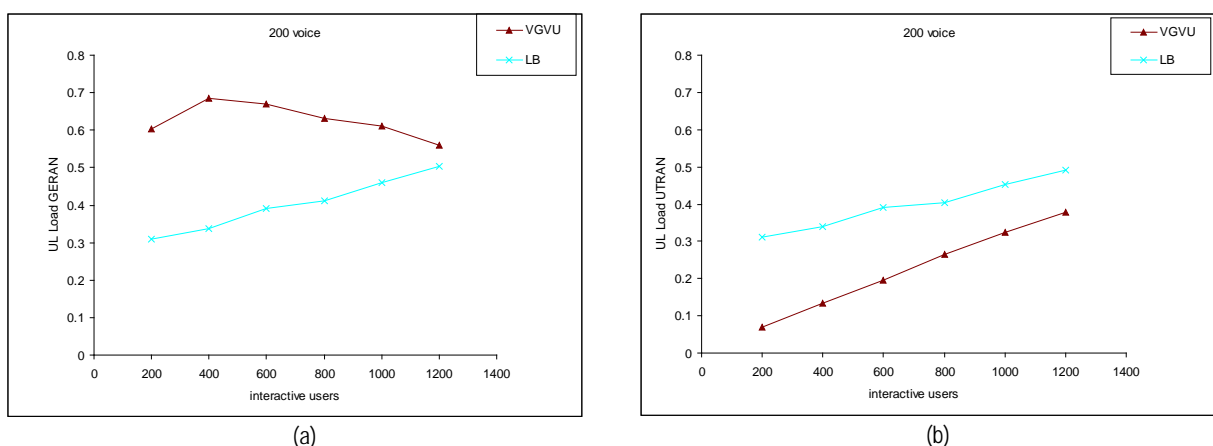


Figure 44- Average UL load in (a) GERAN and (b) UTRAN for policies VGVU and LB with SM2.

In summary, with respect to the initial RAT selection based on load balancing without including VHO, results revealed a tight dependency between the suitability of load balancing RAT selection and

¹¹ In SM1 the number of interactive users is fixed while voice users increase. On the contrary, in SM2 the number of voice users is fixed while the number of interactive users increases

service-class mixing. It has been shown that even though the overall throughput may increase with load balancing policies, this at the expense of interactive traffic performance degradation. Nevertheless, other service type mixings showed no type of throughput improvement at all

In turn, the introduction of VHO capabilities allows higher flexibility in the allocation of multi-service users in a multi-access scenario. We have compared two initial RAT selection policies along with a tight approach for VHO procedures. Results indicate that no remarkable improvement is noted on the total aggregate throughput when using the LB policy as opposed to the VGVU policy. Moreover, with LB, interactive users undergo higher average packet delays which impact the user's perceived QoS. However, we have seen that load balancing procedures may improve the call dropping probability due to a more flexible allocation of users onto both RATs, which is also a key performance indicator to consider.

III- Path Loss - Based RAT Selection

In FDMA/TDMA-based access systems (e.g. GSM/GPRS) there is no intra-cell interference, whereas in contrast, in CDMA-based systems (e.g. UMTS) the intra-cell interference is caused by every single user transmitting in the cell. Furthermore, inter-cell interference is also originated by all simultaneous users in all neighbouring cells, since a complete frequency reuse is considered. CDMA systems are much more sensitive to multi-user interference than FDMA/TDMA ones. The underlying idea of the CRRM approach developed here is to take advantage of the coverage overlap that several RANs using different access technologies may provide in a certain service area in order to improve the overall interference pattern generated in the scenario for the CDMA-based systems and, consequently, to improve the capacity of the overall heterogeneous scenario. This can be achieved by controlling the effective cell radius of CDMA-based systems (i.e. a controlled cell-breathing effect) through appropriate RAT selection approaches that take into account the measured path loss. In this way, the interference level in CDMA-based RATs is reduced while at the same time the target coverage area is assured by means of the cooperation of the FDMA/TDMA-based RATs. The above concept, denoted here as Network-Controlled Cell-Breathing (NCCB) can be effectively complemented with load balancing considerations.

Figure 45 presents the comparison in terms of the total aggregated throughput in the scenario (i.e. including UTRAN and GERAN) against a load balancing (LB) approach when only voice service is considered. The benefits of the proposed strategy can be clearly observed with throughput increases of around 13% in the uplink and 24% in the downlink. Similarly, Figure 46 illustrates the improvements obtained in terms of dropping and blocking probabilities, respectively, showing the better efficiency of the NCCB strategy to allocate the available resources in the two RATs among the different users.

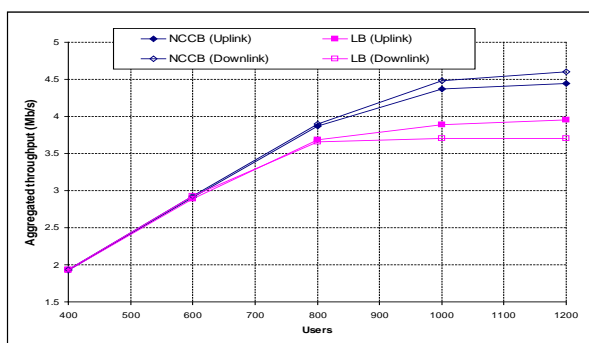


Figure 45.- Total throughput in uplink and downlink

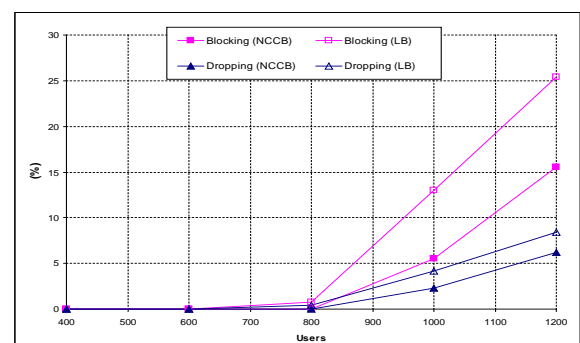


Figure 46.- Blocking and dropping probabilities

In turn, Figure 47 and Figure 48 show the effect of varying the main parameter of the NCCB algorithm, which is the threshold PLth (i.e. users with path loss below PLth are allocated in UTRAN while users with path loss above PLth are allocated in GERAN). It can be observed that the value PLth=120 dB, corresponding to the 60-th percentile of the path loss distribution (and therefore achieves a certain degree of “load balancing” between the two RATs) obtains the best performance. Notice in any case that the improvement with respect to LB comes from the smarter way of distributing users provided by NCCB.

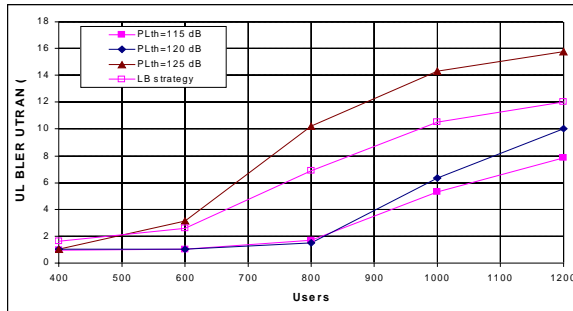


Figure 47.- UL BLER in UTRAN for different values of the threshold PLth

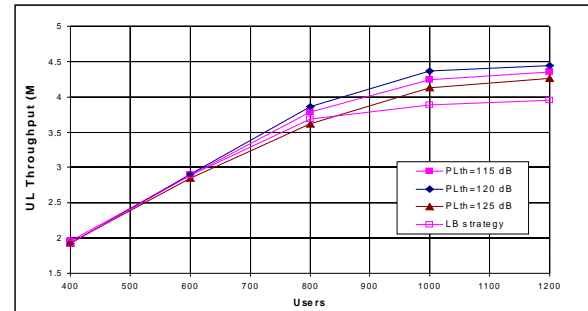


Figure 48.- Uplink throughput for different values of PLth

In summary, when considering a single voice service, the NCCB strategy has been compared against a classical load balancing strategy that tries to keep the same load level in both RATs. Results reveal that a significant improvement in terms of capacity for both uplink and downlink is achieved with the proposed strategy. It has been also shown that, by a proper setting of the maximum path loss PLth, load balancing principles can also be achieved in NCCB, thus obtaining the benefits in terms of flexibility of load balancing while at the same time exhibiting a higher capacity than a pure load balancing.

When considering a mix between voice and www users, different combinations of the NCCB strategy with service-based policies have been tested. It has been observed that the best performance is achieved with the so-called NCCB_voice, corresponding to applying the NCCB strategy only to voice users and allocating www users in UTRAN according to the service-based policies. It has been also observed that the adequate setting of the threshold PLth depends on the existing traffic mix and the trade-off among www delay and voice BLER.

IV- RAT Priority List-Based RAT Selection

The considered algorithm intends to distribute the traffic between the different RATs following several targets and makes use of service-based policies in the form of a RAT priority list for each specific service, while at the same time it introduces load-balancing concepts.

The objective of the study was the analysis of the possible impact in network capacity of a joint admission control 2G/3G which takes into account the kind of service. The strategies simulated are listed below:

- **Strategy A:** No control over admission control is applied. When users may be served by both technologies, GERAN and UTRAN, the decision is taken in a random way, without no predefined decision from the the operator. That is, both RATs have the same priority for all the services.
- **Strategy B:** Voice service users are only served by GERAN. Data services users are served by UTRAN if is possible; if not, they are served by GERAN.
- **Strategy C:** Voice service users are served by GERAN if is possible; if not, they are served by UTRAN. Data services users are served by UTRAN if is possible; if not, they are served by GERAN.

Figures 49 and 50 show the percentage of attended users for both voice and video streaming services. Similar results are obtained for video-telephony and browsing services

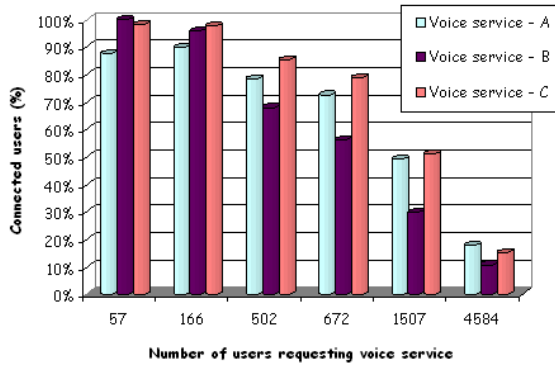


Figure 49.- Percentage of attended users for voice service

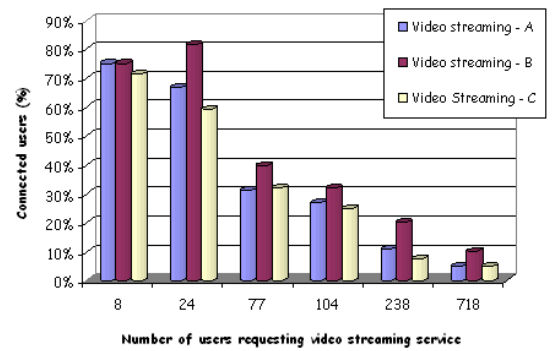


Figure 50.- Percentage of attended users for video streaming service

From the results of the simulations the following conclusions may be inferred:

- For low load situations, the strategy B (voice service only provided by GERAN and data services for UTRAN if is possible and, if not, for GERAN) shows the better performance for all the services.
- For high load situations, the strategy C shows the better behaviour in general because the network capacity is maximized in the work conditions (strong presence of voice service in the service mix). In this situation, the strategy B is which shows the worst performance with a high number of voice users not served. However, it is possible that strategy B would perform better in a scenario with a strong presence for data services

V- Perceived TCP Throughput in CRRM Framework

The impact of CRRM strategies on TCP throughput has also been studied in a scenario including 3G with HSDPA, 2G and WLAN systems. Four different CRRM algorithms, described in table 7, have been evaluated and compared against the manual RAT selection algorithm.

Table 7.- Four different CRRM algorithms used for CRRM performance evaluations

	Type of algorithm	Algorithm description
Algorithm 1	Long term CRRM optimisation criteria	At each evaluation the CRRM decisions are made trying to minimise the time to empty all buffers assuming the current load.
Algorithm 2	Short term CRRM optimisation criteria	At each evaluation the CRRM decisions are made trying to minimise the time until the first data session is finalised.
Algorithm 3	RAT prioritisation for CRRM	According to service prioritisation due to SLA or other operator specific QoS strategies, the service with highest priority is allocated to the RAT with highest system capacity, i.e. WLAN in this scenario. If the high priority data would overload the high capacity RAT, the next RAT (e.g. UTRAN) is also filled with high priority data etc. If the system can handle more traffic, the payload with the second highest priority is allocated in the CRRM using the same principle, i.e. with WLAN usage as best choice.
Algorithm 4	Manual RAT selection	The manual RAT selection used is that all payload with highest priority is allocated to WLAN, while the lowest priority payload is allocated to GERAN. The payload from other "middle priority services" is then allocated to UTRAN. This CRRM algorithm is used as the benchmark for the other algorithms.

The obtained results show that without HSDPA, throughput increases up to 40% have been observed thanks to CRRM. When introducing HSDPA capabilities, the improvement in 3G throughput leads to increases of up to 60%.

VI- Impact of Multi-Mode Terminals on CRRM Performance

Today's wireless communications comprise a broad variety of Radio Access Technology (RAT) standards. In order to take full advantage of B3G networks, existing mobile terminal capabilities need to be extended. Particularly, to provide connectivity to a variety of underlying access technologies is a

must. Therefore, the impact of multi-mode terminals in the framework of heterogeneous networks with an initial RAT selection policy has also been assessed.

Figure 51 shows the throughput degradation defined as the throughput decrement with respect to the throughput achieved when 100% of users have multi-mode terminals in a scenario where RAT selection is done according to the service-based policy VG*VU. Notice that, for different multi-mode terminal availabilities and different number of users requesting service, throughput degradation exhibits different trends. In particular, for 200 voice and 200 www users, no big differences are noticed, meaning that, in this case, GERAN is able to manage voice and single-mode terminals with ease. Certainly, the average timeslot utilisation factor in GERAN reveals an occupation of resources below 70%. The increase of users requesting to be served is translated into a bigger degradation in terms of throughput as multi-mode terminal availability decreases. While for 400 voice and 400 www users degradation starts to get noticeable for multi-mode availabilities of 25% and 50%, for 600 voice and 600 www users this degradation is already perceptible at 75% of multi-mode availability.

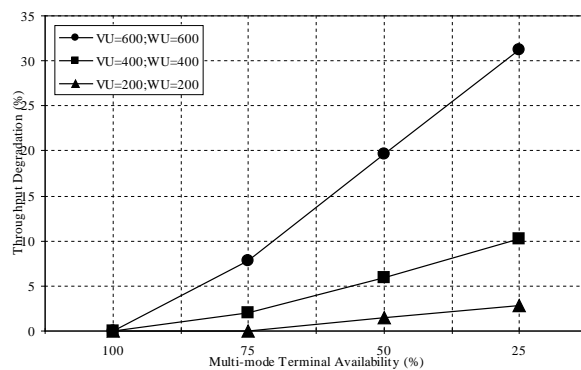


Figure 51.- Uplink throughput degradation due to multi-mode terminals

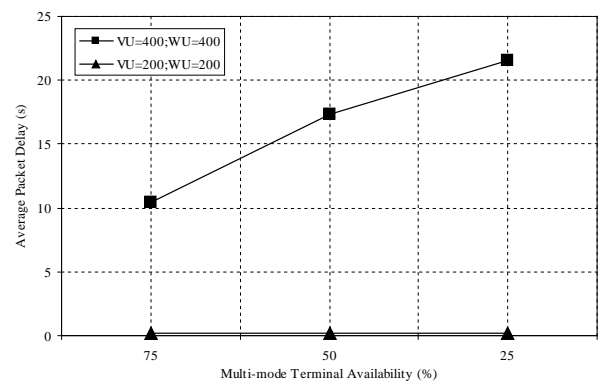


Figure 52.- Uplink average packet delay for interactive users.

Packet delay statistics for interactive users also reveal degradation when considering a scenario with mixed multi-mode and single-mode terminals. This degradation impacts directly in the perceived QoS by the data user and it is therefore important to keep its value as low as possible. Figure 50 shows the uplink average packet delay for interactive users being served through GERAN for different mixings of multi-mode terminals and service-classes. Notice the increasing packet delay when multi-mode terminal availability decreases for 400 voice and 400 www users. As for 200 voice and 200 www users, the average packet delay remains almost constant with multi-mode terminal availability and also at an acceptable level, therefore exhibiting no degradation in this sense.

The suggested approach to overcome the delay degradation resides in the introduction of dedicated EGPRS timeslots for interactive users. In this way, some resources may be reserved for interactive users and therefore packet delay performance can be improved with respect to not having any reservation scheme. As for throughput performance, it can be foreseen that contribution of voice users to the total aggregate throughput might diminish. This reduction of voice throughput may be compensated, given certain conditions, by interactive users allocated to the reserved slots. This entails a trade-off between the number and the applicability of reserved resources (slots) for interactive users in GERAN and other parameters, such as offered load and multi-mode terminal availability, as it is observed in the following table for different numbers of voice users (VU) and web users (WU).

Table 8.- EGPRS Slot Reservation Gain (%).

$\bar{u}_i = (VU_j, WU_j)$		Multi-mode Terminal Availability (%)			
VU _j	WU _j	25	50	75	100
200	200	0.00	0.00	0.72	0.00
	400	1.14	0.56	1.13	0.55
	600	0.00	0.46	-0.46	-0.91
400	200	-3.88	-4.76	-6.07	-7.83
	400	3.49	-0.42	-4.40	-5.10
	600	8.71	2.21	-1.39	-4.71
600	200	1.38	-2.10	-2.75	-1.48
	400	13.22	7.00	-0.35	-4.23
	600	13.87	12.23	3.45	-1.16

In summary, the obtained results indicate degradation in terms of throughput introduced by the limited operation of single-mode terminals. By considering a reservation scheme in GERAN for interactive users, we can improve the average packet delay for such users. While for a high multi-mode terminal availability results indicated that the reservation scheme was not necessary, for lower multi-mode terminal availabilities this scheme improved both aggregated throughput and packet delay figures, particularly for high number of interactive users.

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4.3 EVEREST Testbed

The EVEREST demonstrator (testbed) has been used to validate in the laboratory the benefits of the proposed RRM/CRRM algorithms and QoS Management techniques.

The EVEREST demonstrator is a real-time operation HW/SW platform including multimedia terminals, GERAN/UMTS/WLAN elements and IP based Core Network, able to support real-time multimedia calls. The purpose of the GERAN/UMTS/WLAN emulators is to reproduce the real behaviour of a user under test with more accuracy than what would be obtained from the simulators (more suited for global systems performance analysis), but with less implementation complexity than in a real system. The philosophy used is to implement a "subset" of functionality, appropriate for emulation of the critical aspects related to RRM/CRRM and QoS issues, rather than to realise a "one by one" representation of their specs. This approach leads to a lighter implementation, suitable to assess new RRM/CRRM algorithms easily, not considering parameters that do not have a relevant impact on them. That is, the EVEREST demonstrator is understood as a flexible SW/HW platform that allows the experimental evaluation of new Radio Resource Management algorithms under controlled but realistic conditions

Testbed Architecture Overview

From a conceptual view point the testbed architecture can be divided in two main blocs: the Heterogeneous Radio Access part and the Core Network.

Radio Access Domain

The functional architecture of the proposed approach to emulate a heterogeneous RAN is shown in the following figure.

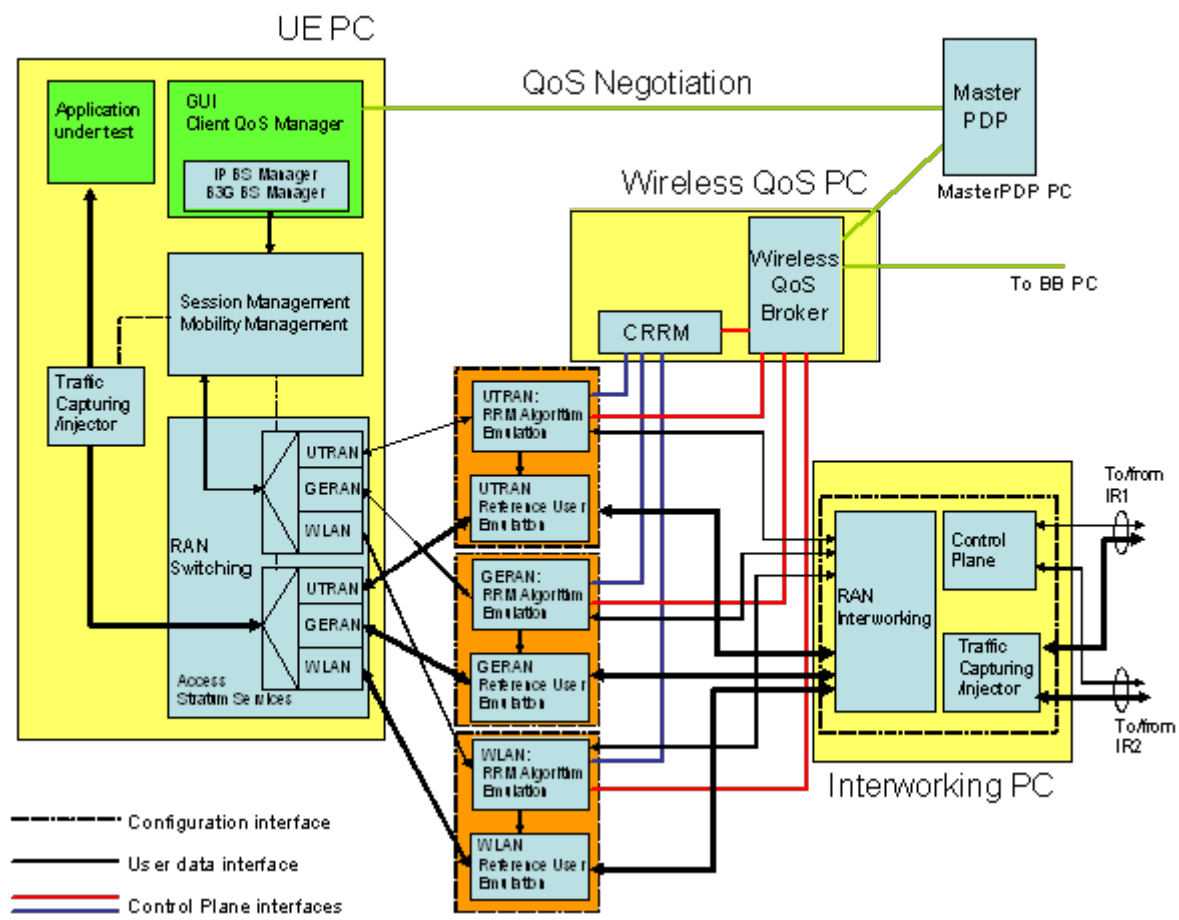


Figure 51.- Heterogeneous RAN Emulation approach

Functions identified are the following:

- **Client QoS Manager.** Module in charge of QoS negotiation. Through a Graphical User Interface (GUI) this module will provide the capabilities to ask for QoS in the terms to be demonstrated under EVEREST. Following the policy-based framework introduced in UMTS R5, QoS negotiation is intended to be handled at service level through a Policy master Decision Point element in charge of QoS provision in the whole access network (RAN and CN).
- **Session and Mobility Management.** Module in charge of providing Non Access Stratum services as Session and Mobility management. Particularly, management of PDP contexts will be performed in this module.
- **Traffic Capturing/Injector Module.** Real IP packets coming from /going to the application under test are captured in the corresponding network interfaces and delivered at user plane to the radio processing modules. This module will be configured based on negotiated Traffic Flow Templates (TFT).
- **RAN Switching Module.** Ad-hoc module needed to deliver the user data and signalling traffic to the currently connected RAN. This module will be managed at run-time by the control entities (Session Management).
- **Wireless QoS Broker.** Two basic functionalities are envisaged for this module:
 - CRRM functionalities
 - QoS Management in the Heterogeneous RAN.
- **Interworking Elements.** Its main functionality is to allow different coupling configurations to be tested. In this sense such element are useful to configure and manage, at the initialising phase of the testbed or during some kinds of handover the current links among each RAN and the Ingress Routers of the CN (therefore this element allows higher levels of flexibility to test different scenarios, e.g., all the RANs connected to the same SGSN, GERAN and UTRAN connected to a SGSN and WLAN to a different one, etc.).
- **RAN Emulation Modules.** Their main functionalities tries to reproduce under a simplifying approach the relevant behaviour, in terms of its influence to the user data traffic, of the different Radio Access Technologies considered.

Core Network Domain

For the core network part, there is no emulation. The tests are carried out using the communication stack of the Linux operating system, which acts as an IP router. In Figure 52, the topology of the IP CN and the IP mobility entities can be seen. As it can be seen in Figure 52, the CN testbed can also be seen as a standalone testbed where WLAN 802.11b cards are used for the wireless interface. Thus, tests can be carried out separately before the integration of the radio and the IP CN parts.

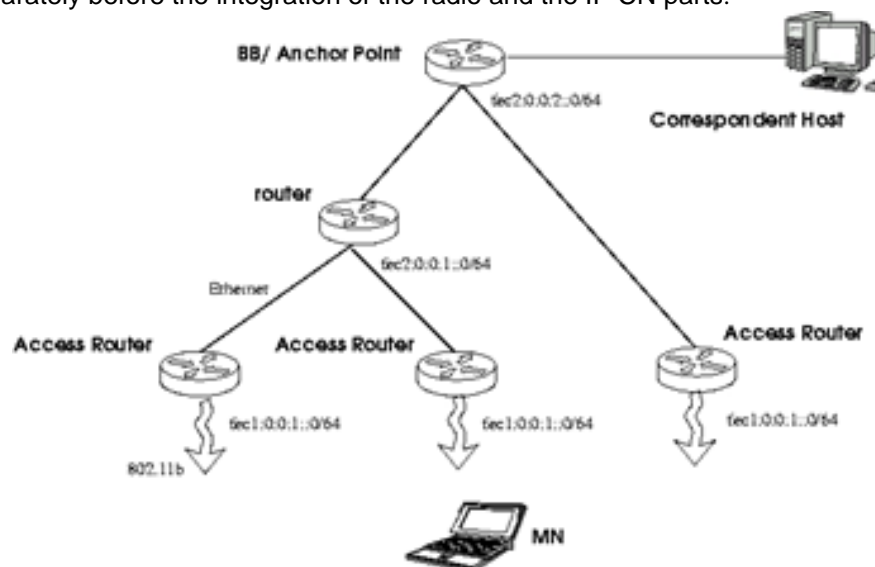


Figure 52. CN testbed layout.

On all the nodes the Linux kernel 2.6 is used. The routers are configured statically; no dynamic routing daemon is used. In order to implement the DiffServ data-plane, the Linux traffic control is used. The Queueing discipline chosen is the Hierarchical Token Buckets with three DiffServ code-points: AF, EF, BE. All routers have a default DiffServ setup, which is configured manually before tests thanks to the traffic control command line "tc" and the IP filtering tool "iptables". Only the edge routers, which here correspond to the Access Router (AR) and gateway, are configured dynamically by the BB entity. The testbed platform is composed of two programs: one which implements the BCMP micromobility management and the other which implements the functionalities of a policy-based BB.

The logical connections of the CN entities can be seen in Figure 53. Moreover in this figure, the control and the data plane are distinguished: the data plane of DiffServ can be separated into the edge router part and the core router part; the control plane of DiffServ is represented by the BB, and regarding the mobility management only the control plane is represented.

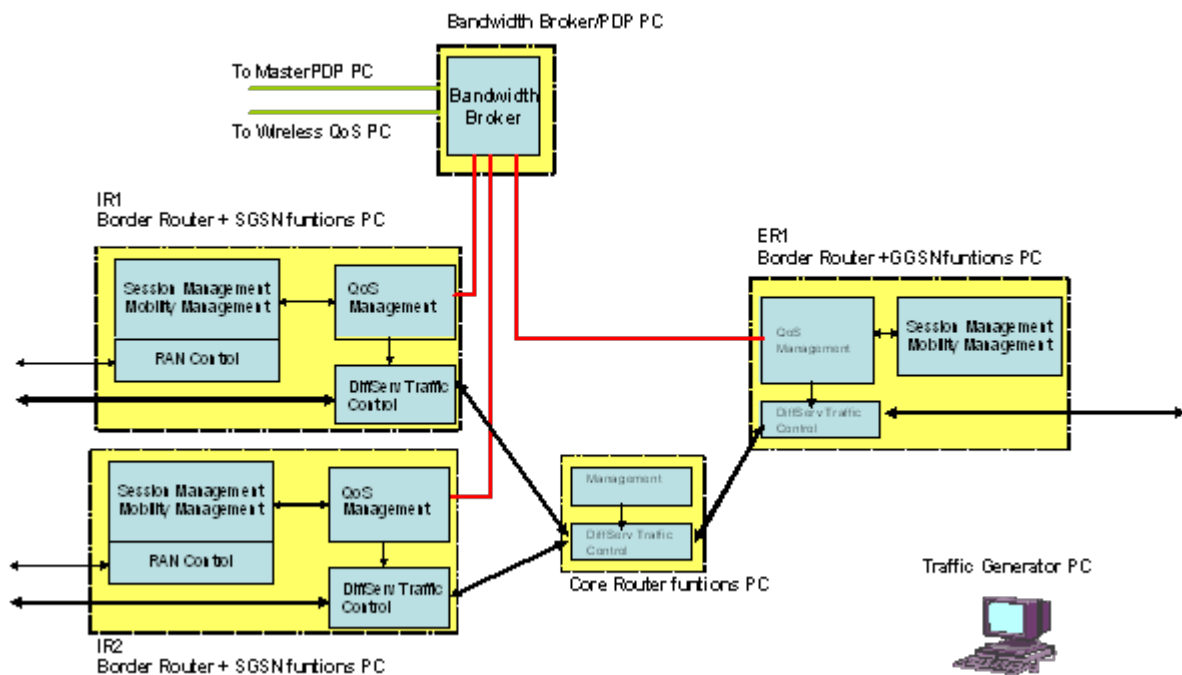


Figure 53.- Logical connections of the CN entities

The functions identifies are the following:

- **Edge Routers (Ingress Routers-IR, Egress Route-ER)** The functionalities in charge of the edge routers are twofold: QoS and mobility management. Regarding QoS, the edge router has to classify and give different forwarding treatment to incoming packets. Regarding IP mobility management, the edge router is in charge of the packet encapsulation and decapsulation
- **Core Router.** A core router has only the functionalities of an IP router, enhanced by DiffServ forwarding functions. It does not have any interface with control management (BB, anchor point).
- **Bandwidth Broker.** The BB controls the network resources, by giving different priorities to aggregated flows in the DiffServ domain and also interacts with QoS Managers (MPDF and WQB) for negotiation and reservation.
- **MPDF.** A Master PDF is the element in charge of the overall end-to-end QoS management, then interacting with WQB for the radio part and the BB for the core network part.

The physical architecture of the testbed is show in the following picture and reflects the functional architecture described in the previous section.



(a)



(b)

Figure 54- EVEREST Testbed.

- (a) The PC placed right side is the user equipment, whereas the PC placed in the left side is the Graphical Management Tool
 (b) The PCs allocated in the rack placed in the right side corresponds at the implemented Core Network part, whereas the PCs allocated in the rack placed in the left side emulates the different Radio Access Technologies.

Software Environment

Operating System

The operating system selected for all the PCs in the testbed is Linux with a kernel 2.6.x. Any Linux distribution is suitable since required features mostly rely on kernel and not on installed software for each distribution. In particular, the testbed machines appearing in Figure 54 have in rack 2 the Linux kernel 2.6.7 and in rack 1, UE and Server the Linux kernel 2.6.3. In the case of the UE and Server, the installed distribution is Fedora Core (release 3), the sequel of Red Hat after version 9.

Communications Manager

Communications Manager (CM) is a software tool mainly devoted to integrate software from different developers and manage its execution on a networked cluster of PCs with a Linux operating system. An application under CM is made of software modules running in parallel that are joined through interfaces adequately matched. It also offers means to such software to interact with the controlling entity of the system by means of dynamically modified parameters and statistics. Finally, CM controls the execution of the software in a slotted temporal framework to provide to the application the required timing.

Application in the testbed

The development of applications was not an objective of the EVEREST project. Therefore, the applications used have been chosen taking into account the services that they are intended to cover. Also, the costs associated for obtaining these applications have been taken in consideration; common freeware applications were preferential. For that reason, at least one application per service class: conversational, streaming, interactive or background was selected.

Testbed Management Capabilities

The following management capabilities are supported in the EVEREST testbed

- Control the execution flow of the testbed (init, run, pause, restart options through CM facilities) and selection of the scenario to be demonstrated. The scenario here is understood as the needed software modules and their specific configuration involved in the demonstration of certain procedures or capabilities.

- Configuration of all the initialisation parameters required in the modules running in the test bed. Parameters needed to initialise the different software modules can be set-up centrally from the management platform.
- Collect and correlate logged data from the different modules of the demonstrator. Traces generated by the different modules share the same format in order to make easy the integration of the data into a single file. Furthermore, a tool is available to post-process such traces allowing replaying the logged events in a dynamic way.
- Observe statistics during the execution of a demonstration (on-line representation/visualization).
- Change some configuration parameter during the execution of a trial to force a given situation (i.e. increase the number of users dynamically to analyse consequences over radio bearers established by the user under test).

The management platform developed during the ARROWS project and referred to as Arrows Graphical Management Tool (AGMT) has been upgraded in order to include aforementioned functionalities in the EVEREST testbed. Under the testbed architecture the AGMT software should be installed in a separated PC, which must be connected to the public LAN of the whole testbed. The software of the AGMT has been developed in JAVA due to its facilities in programming graphical and networked applications and to allow the portability to different Operating System platforms. The acronym AGMT for the management tool has been left unchanged in EVEREST although it now stands for Advanced Graphical Management Tool to account for the modifications carried out within this project.

Remote testbed Capabilities

The EVEREST testbed is being designed for remote operation. In particular, the two main capabilities under consideration are the following:

- Remote testbed management and monitoring. This capability implies the remote operation of the testbed by accessing to the testbed management and monitoring tools. In this way, the testbed can be configured and the emulation of a given scenario controlled remotely from a machine outside the UPC premises. Nevertheless, applications of the user under test are locally executed in the testbed machines.
- Remote Execution of Applications. Applications tested over the testbed are executed in external machines other than the UE and Server machines in the UPC premises. So, traffic from a remote location is routed towards the testbed, passed through the testbed data path and then forwarded to its destination that could be located remotely as well. This usage is intended to test applications developed over specific platforms that can not be ported easily to the testbed machines. Figure 55 shows the testbed configuration for the remote execution of applications.

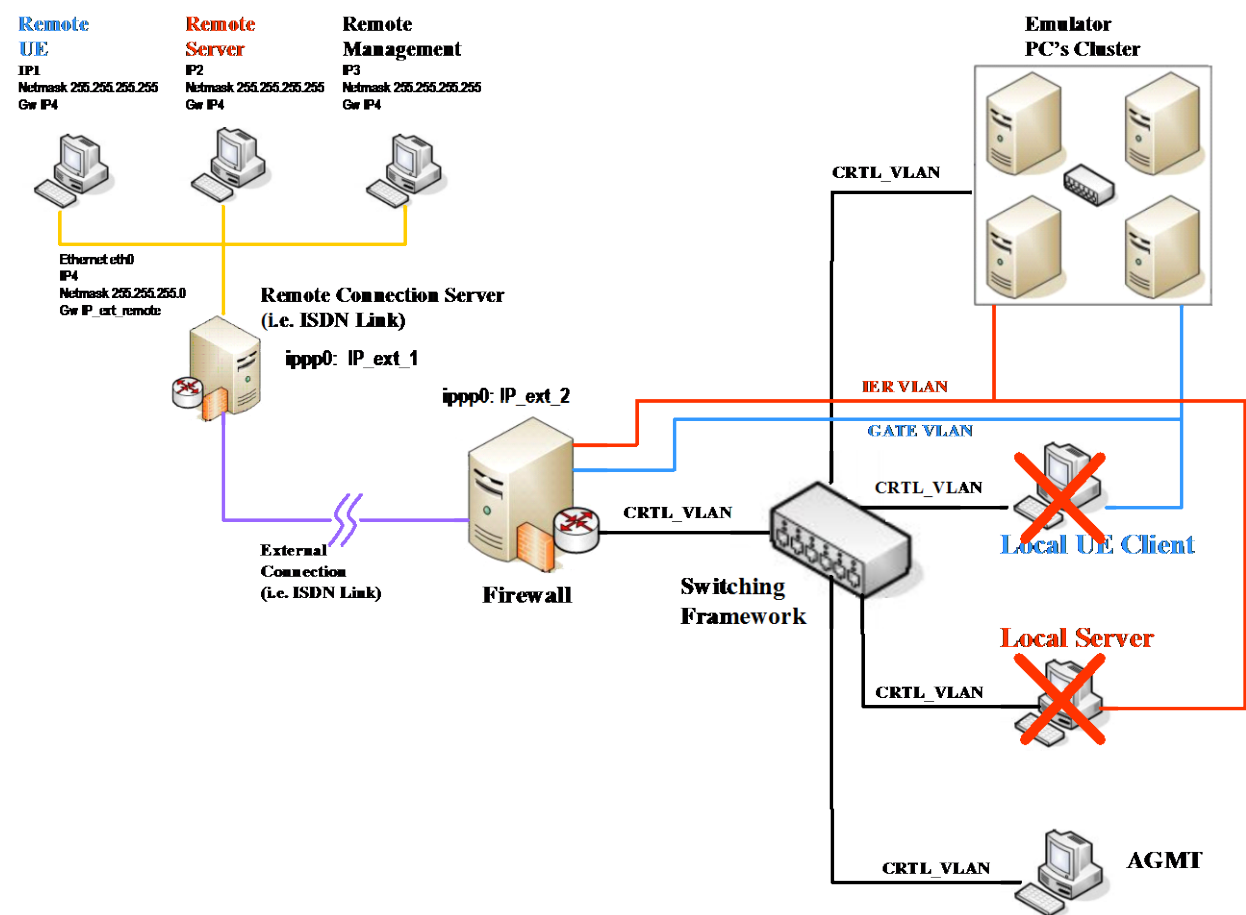


Figure 55. Testbed Configuration for the remote execution of applications

5 **Main conclusions reached**

The **overall conceptual studies and the implementation** of the testbed have given good knowledge on the Radio Resource and QoS Management techniques. Complexity, feasibility and impacts of the selected solutions have produced a positive impression.

From a macro prospective, we have derived the following main lessons:

The RRM/CRRM is a complex problem with many factors influencing in the achieved performance and with many mixing effects. The intensive simulation work carried out in EVEREST has provided a solid background and a significant engineering enrichment. The step by step approach that has been followed, starting with the simpler scenarios and ending up with the more involved ones, has been revealed very useful to cope with the RRM/CRRM problem. At the end recommendations on:

RRM issues for UMTS

A large variety of studies related to RRM for UMTS have been carried out. Specifically EVEREST has addressed at:

- A new framework for capturing coupling among cells based on load gradient computations.- This framework is presented in a compact formulation for both uplink and downlink, and it allows implementing mechanisms supporting smart load control actions including admission control, congestion control algorithms.
- Indoor Traffic.- The higher power levels needed for indoor service will lower cell capacity both for uplink and downlink traffic.
- Traffic Hot-Spots.- In order to assure the user QoS (Quality of Service) requirements in a hotspot, not only network planning but also RRM algorithms must be considered. In that sense, to adjust the transmission pilot power of the hotspot cells and its adjacent cells could be used to improve the system performances
- Static Traffic.- The flexibility in the provision of multiple bit rate services in 3G communication systems will allow users to benefit from services better adjusted to their specific requirements. In particular, users that receive data traffic services, typically with laptops in scenarios like offices, airports, etc., use to be static or, at least, with a very limited mobility. This fact gives room to propose more sophisticated admission control strategies, which may provide significant performance improvements. Under this framework, an advanced algorithm, denoted as PLEBAC (Path Loss Estimation Based Admission Control), that improves the overall performance while at the same time it guarantees the quality of the accepted users has been proposed taking into account the benefits from the easier predictability in terms of power consumption of static data users.
- Repeaters.- The repeaters are equipments between the radio base station and the terminals, able to amplify the received signals both on the uplink and on the downlink. The possible usage situations for repeaters in a WCDMA system are the following: coverage extension, capacity extension, soft-handover region reduction. In terms of capacity the performance evaluation studies have demonstrated that the percentage of served users in the system increases with repeater gain values lower than 70-80 dB, when the maximum capacity value is obtained.
- Multiple RF Carriers.- The scenario simulated consist in an indoor traffic Hot Spot within an urban area. The UMTS operator uses two frequencies, where frequency f2 only applies at a three sector site and frequency f1 is applied at all the macro layer of cells. The results indicate that for packet switched services (PS), its blocking probability is sensitive to the load allocation on the frequencies, whereas speech blocking and dropping, and PS bit rate, is less sensitive to different load distributions.

- Hierarchical Cell Structures (HCS).- The load gradient framework has been proposed as a candidate to state whether or not is possible to operate different HCS layers with the same carrier frequency depending on spatial traffic distributions.
- Transport Channel Type Switching.- Notice that In the case of a discontinuous data transfer, it is very important to be able to optimize the usage of dedicated transport channel (DCH), preventing channelization code shortage in the downlink without degrading the end-to-end quality of service experienced by the user in an appreciable manner. The achieved results show that the UTRAN capacity is higher when the transport channel type switching algorithm is active.
- High Speed Downlink Packet Access (HSDPA).- Advanced schedulers, that combine both channel information and service requirements, have been evaluated and the results were presented in terms of satisfied users and cell throughput both for indoor and vehicular environments. Comparing the different schedulers proposed in the literature, it was obtained that Priority Standard scheduler provides the better results both for QoS bit rate and satisfied users. Moreover, HSDPA is capable of supporting various QoS levels. In fact, the HSDPA specification supports up to 16 priority levels and 8 queues per user. Then, when heterogeneous traffic coexists, an innovative HSDPA packet scheduler that discriminates QoS classes was proposed to provide better QoS support than those that consider only the channel condition. The simulation results further showed that the proposed scheduler is robust to changes in the traffic constitution, and supports call arrival rates higher than the obtained with conventional schemes.
- RAN Sharing.- Sharing spectrum can be very attractive, for example, in rural areas UMTS coverage can be offered with much lower investment costs, but also in urban areas and hot spot areas capacity gains can be achieved. Certainly, there is a capacity gain due to the increased trunking efficiency as channels are pooled together between the operators. When analysing different approaches, the obtained results show that the so called Multi-cell algorithm provides the best Grade of Service, because it provides higher admission probability than the Half power algorithm and maintains the dropping probability in a low value.
- Location Aware Resource Reservation.- Recent developments in the positioning technology in the context of WCDMA systems provide strong assurance that accurate position measurements. This can be exploited to develop more advanced RRM (Radio Resource Management) strategies that increase the system efficiency. In particular an innovative reservation algorithm, whose main objective is to assure service to business users moving along a main road while at the same keeping the service of consumer users at a satisfactory level, has been proposed. The reservation of certain resources for handover users reduces the number of dropped connections at expense of certain increase in the blocking probability of new connection requests. The proposed algorithm takes advantage of the predictability in the movement of users along a main road in order to determine the most adequate instant of time when the resource reservation for handover users should be made

CRRM issues

Common Radio Resource Management (CRRM) refers to the set of functions that are devoted to ensure an efficient use of the available radio resources in heterogeneous networks scenarios by means of a proper coordination between the different radio access networks. Then:

- With respect to Common Radio Resource Management, a first contribution of the project has been the outline of a functional model for having a common management of the pool of radio resources in heterogeneous scenarios
- CRRM performance evaluation reported here assume a functionality split in which the CRRM takes charge of the RAT selection procedures, including both initial RAT selection and vertical handover, while the RAT-specific RRM algorithms, like admission control, congestion control or packet scheduling are executed locally at the RRM entities.
- With respect to RAT selection a framework for devising policy based strategies has been proposed. They consider both service-based criteria (e.g. allocate voice in GERAN and interactive services in

UTRAN) and radio network-based criteria (e.g. allocate indoor users in GERAN and interactive services in UTRAN) in the form of the so-called complex policies.

- For service-based RAT selection policies and considering scenarios with a mix of indoor and outdoor users with different services, up to medium voice loads (i.e. 200 users) no relevant differences between the policies are observed. However, when the load increases as first option is better to allocate the voice users in GERAN and the interactive (non-real time) users in UTRAN. If there is not more resources available in GERAN then apply a policy that allocates the indoor user in GERAN and the outdoor user in UTRAN, moving voice outdoor user from GERAN to UTRAN.
- With respect to vertical Handover (VHO), two different approaches, denoted as loose and tight inter-working with horizontal HO, have been analysed. It has been shown that the tight approach allows a better fulfilment of the RAT selection policy at expense of an increment in the signalling.
- Load balancing (LB) is another possible guiding principle for resource allocation in which the RAT selection policy will distribute the load among all resources as evenly as possible. When considering LB without VHO procedures, the obtained results revealed a tight dependency between the suitability of load balancing RAT selection and service-class mixing. However, the introduction of VHO capabilities allows improving the flexibility achieving lower dropping ratios than with the service-based policies.
- For Path Loss - Based RAT Selection, the underlying idea of the CRRM approach developed is to take advantage of the coverage overlap that several RANs using different access technologies may provide in a certain service area in order to improve the overall interference pattern generated in the scenario for the CDMA-based systems and, consequently, to improve the capacity of the overall heterogeneous scenario. The above concept has been denoted as Network-Controlled Cell-Breathing (NCCB).
When considering a single voice service the obtained results reveal that a significant improvement in terms of capacity for both uplink and downlink is achieved. When considering a mix between voice and www users, the best performance is achieved with the so-called NCCB_voice, corresponding to applying the NCCB strategy only to voice users and allocating www users in UTRAN according to the service-based criterion.
- When the impact of Multi-Mode terminals on CRRM performance is analysed the obtained results show that degradation in terms of throughput appears but, considering a reservation scheme in GERAN for interactive single-mode users, the average packet delay for such users can be significantly improved.

Annex 1.- List of Publications

The scope of this sub-section is to present the dissemination activities carried out by the EVEREST project in scientific fora. In total the partners have published 32 scientific papers and contributed with 10 presentations in workshops. The publications are ordered according the media used: scientific journals and magazines, international conferences, workshops.

Scientific Journals/Magazines

- Nima Nafisi, A.Hamid Aghavami, "IP handover delay for host-based micromobility protocols" IEE Electronics letters, Electronics Letters; Volume 41, Issue 3, 3 Feb 2005 Page(s):160 - 161.
- Vasilis Friderikos, Lin Wang, Mikio Iwamura, A. Hamid Aghvami, "Color-aware Power and Rate Adaptation in IP-based CDMA Radio Access Networks", Journal of Wireless Communications and Mobile Computing, Volume 5 , Issue 4, June 2005 pp 407-420
- L. Wang, A.H. Aghvami, W. G. Chambers, J. Perez Romero, O. Sallent, "Performance Analysis of an Integrated CS/PS Services CDMA System", IEEE Transactions on Vehicular Technology, Volume 54, Issue 4, July 2005 Page(s):1488 – 1499

Conferences:

The following papers were presented in International Conferences:

- F. Adelantado, O. Sallent, J. Pérez-Romero, R. Agustí ; "Impact of Traffic Hotspots in 3G W-CDMA Networks " the IEEE Vehicular Technology Conference 2004 Spring. Milan (Italy), 17-19 May, 2004
- F. Casadevall; P. Karlsson; O. Sallent; H. Gonzalez; A. Barbaresi; M.Dohler; M.Dinis; " EVEREST Overview ", IST Summit 2004. Lyon (France), 28-30 June, 2004.
- N. Nafisi; L. Wang, H.Aghvami, R. Ferrus, A. Gelonch, J. Perez-Romero, O. Sallent, R. Agusti; "Extending QoS Policy-based mechanisms to B3G Mobile Access Networks "; IST Summit 2004. Lyon (France), 28-30 June, 2004.
- J. Perez-Romero, O. Sallent, D. Ruiz, R. Agustí; "Admission Control Algorithm to Manage High Bit Rate Static Users in W-CDMA"; IST Summit 2004. Lyon (France), 28-30 June 2004.
- Lin Wang, Vasilis Friderikos, A.Hamid Aghvami, Misha Dohler; "Color-aware Link Adaptation for DiffServ over CDMA Systems"; IST Summit 2004. Lyon (France), 28-30 June 2004.
- J. Pérez-Romero, O. Sallent, R. Agustí; "Impact of Indoor Traffic on W-CDMA Capacity "; PMIRC04. Barcelona (Spain), 5-8 September 2004.
- J. Perez-Romero, O. Sallent, R. Agusti, "On the Capacity Degradation in Uplink/Downlink W-CDMA Due To Indoor Traffic"; IEEE Vehicular Technology Conference 2004 F all. Los Angeles (USA), 26-29 September 2004.
- J.L. Valenzuela, A. Monleon, I. San Esteban, M. Portoles, O. Sallent; "A Hierarchical Token Bucket Algorithm to Enhance QoS in IEEE 802.11: Proposal, Implementation and Evaluation", IEEE Vehicular Technology Conference 2004 Fall. Los Angeles (USA), 26-29 September 2004.
- J. Pérez-Romero, O. Sallent, R. Agustí , " A Novel Approach for Multicell Load Control in W-CDMA "; 5th International Conference on 3G Mobile Communication Technologies (3G 2004), London (UK), 18-20 October, 2004.
- J. Pérez-Romero, O. Sallent, R. Agustí, "On Evaluating Beyond 3G Radio Access Networks: Architectures, Approaches and Tools", IEEE Vehicular Technology Conference 2005 –Spring, Stockholm, Sweden, May 30 - June 1, 2005

- J. Sánchez-González, O. Sallent, J. Pérez-Romero, R. Agustí. "An Admission Control Algorithm for WCDMA Considering Mobile Speed and Service Characteristics", IEEE Vehicular Technology Conference 2005 –Spring, Stockholm, Sweden, May 30 - June 1, 2005
- R. Ferrús, A. Gelonch, J. Pérez, O. Sallent, N. Nafisi, M. Dohler, "A feasible approach for E2E QoS management in coordinated heterogeneous radio access networks", 24th IEEE International Performance, Computing, and Communications Conference, 2005. (IPCCC 2005); 7-9 April 2005.
- M. Iwamura, V. Friderikos, L. Wang, H. Aghvami; "Biased Adaptive Modulation/Coding to Provide VoIP QoS over HSDPA", 2005 IST Mobile Summit Dresden (Germany) 19-22 June, 2005.
- Barbaresi, S. Barberis, P. Gorla; "Admission Control Policy for WLAN Systems based on the "Capacity Region", 2005 IST Mobile Summit Dresden (Germany) 19-22 June, 2005.
- R. Ferrús, A. Gelonch, F. Casadevall, X. Revés, N. Nafisi, M. Dohler; "Testbed for CRRM and E2E QoS Management Evaluation in B3G Networks"; 2005 IST Mobile Summit Dresden (Germany) 19-22 June, 2005.
- R. Ferrús, A. Gelonch, O. Sallent, J. Pérez-Romero, N. Nafisi, M. Dohler; "Vertical Handover Support in Coordinated Heterogeneous Radio Access Networks"; 2005 IST Mobile Summit Dresden (Germany) 19-22 June, 2005.
- X. Gelabert, J. Pérez-Romero, O. Sallent, R. Agustí, F. Casadevall; "Radio Resource Management in Heterogeneous Networks"; 3rd International Working Conference on Performance Modeling and Evaluation of Heterogeneous Networks (Het-Nets 05), Ilkley, West Yorkshire (UK), 18th-20th July 2005.
- Lin Wang, Mikio Iwamura, Vasilis Friderikos, Nima Nafisi and A. Hamid Aghvami; "DiffServ Aware Link Adaptation for CDMA Radio System"; Int. Conf. on Quality of Service in Heterogeneous Wireless/Wired Networks; Orlando-USA; 22-24 August 2005
- X. Gelabert, J. Pérez-Romero, Oriol Sallent, R. Agustí; "On the Impact of Multimode Terminals in Heterogeneous Wireless Access Networks"; 2nd International Symposium on Wireless Communication Systems 2005; Siena- Italy; 5-7 September 2005.
- N. Nafisi, R. Ferrús, A. Gelonch, O. Sallent, J. Pérez-Romero, L. Wang, M. Dohler, H. Aghvami; "QoS Aware Path Selection in a B3G System", PIMRC'05 Berlin (Germany) ; 11-14- September 2005
- J. Pérez-Romero, O. Sallent, R. Agustí, P. Karlsson, A. Barbaresi, L. Wang, F. Casadevall, M. Dohler, H. González, F. Cabral-Pinto; "Common Radio Resource Management: Functional Models and Implementation Requirements", PIMRC'05 Berlin (Germany) 11-14- September, 2005
- J. Sánchez-González, O. Sallent, J. Pérez-Romero, R. Agustí; "On Managing Dynamic Traffic Hotspots in WCDMA Networks"; PIMRC'05 Berlin- (Germany) 11-14- September
- P. Emanuelsson, et al.; "Admission Control at UMTS Spectrum Sharing" ; Int. Conf. on Software , Telecommunication and Computer Network; Split (Croatia), 13-17 September 2005 .
- J. Pérez-Romero, O. Sallent, R. Agustí , "Policy-based Initial RAT Selection algorithms in Heterogeneous Networks" 7th Int. Conf. on Mobile and Wireless Communications Networks 05; Marrakesh – (Morocco) ; 19-21 September 2005;
- X. Gelabert, J. Pérez-Romero, O. Sallent, R. Agustí, " On the Suitability of Load Balancing Principles in Heterogeneous Wireless Access Networks" ; 1st Int. International Wireless Summit; IWS 05. Aalborg (Denmark); 17-22 September 2005
- J. Majkowski, F. Casadevall; "Admission Control in IEEE 802.11e EDCA" ; 1st Int. International Wireless Summit; IWS 05. Aalborg (Denmark); 17-22 September 2005)

- F. Casadevall, P. Emmanuelsson, R. Ferrus, A. Gelonch, P. C. Karlsson, N. Nafisi, J. Perez-Romero, O. Sallent, L. Wang, "EVEREST – A Policy Based QoS Architecture for Heterogeneous Wireless Systems", IST4Balt Conference "Evolving Mobile Europe " Vilnius (Lithuania) 24th and 25th October 2005.
- Lin Wang, Vasilis Friderikos, Mikio Iwamura, Nima Nafisi, A. Hamid Aghvami, Misha Dohler, "A Link Adaptation Scheme to Support Relative DiffServ QoS over CDMA Radio Systems", IEE 3G and Beyond Conference 2005; London (UK) 7-9 November 2005.
- F. Adelantado, F. Casadevall, "Dynamic Common Pilot Power Management in a Real Hot Spot"; IEEE GLOBECOM 2005 St. Louis-Missouri (USA), November 28th –December 2th 2005.

Workshops

The following presentations were done in the Workshops

- R. Ferrus, A. Gelonch, F. Casadevall, J. Pérez-Romero, O. Sallent, R. Agustí, Nima Nafisi, Lin Wang, M. Dohler, Hamid Aghvami, Peter Karlsson; "End-to-End QoS Architecture for Multi-Domain and Wireless Heterogeneous Access Networks: The EVEREST approach". WWRF workshop in Beijing (China), February 2004.
- F. Casadevall "EVEREST: Envisaged Scenarios". Cluster SB3G scenarios workshop, Brussels March 2004.
- J. Pérez-Romero; "Envisaged RRM: Location Aided Approaches in EVEREST". Cluster LOBSTER workshop, Brussels March 2004.
- J. Pérez-Romero, O. Sallent, D. Ruiz, R. Agustí, "PLEBAC: A New Downlink Admission Control for UTRAN-FDD", Mobile Venue'04 Workshop. Athens (Greece) 27-28 May 2004.
- R. Ferrús, A. Gelonch, F. Casadevall, X. Revés, N. Nafisi, "An E2E QoS Testbed for B3G networks", E2R workshop, Barcelona (Spain), 5 September 2004.
- N. Nafisi, L. Wang, M. Dohler, H. Aghvami, R. Ferrús, A. Gelonch, O. Sallent, J. Pérez-Romero, F. Casadevall, R. Agustí, "End-to-end QoS architecture and testbed for B3G systems: the Everest approach", Workshop WRRF – Canada, November 2004.
- J. Pérez-Romero; "Radio Resource Management Strategies in the framework of the EVEREST project", Cluster SB3G: Workshop on Radio Resource Management, Brussels 9th March 2005.
- R. Ferrús, N. Nafisi, A. Gelonch, O. Sallent, J. Pérez-Romero; "EVEREST QoS Architecture", Workshop Trends in Radio Resource Management, Barcelona, 16 November 2005.
- O. Sallent, F. Casadevall, P. Karlsson, A. Barbaresi, L. Wang, H. González, F. Cabral-Pinto "EVEREST'S Activities on RRM/CRRM: an overview", Workshop Trends in Radio Resource Management, Barcelona, 16 November 2005.
- R. Ferrús, A. Gelonch, X. Revés, N. Nafisi, "EVEREST Testbed", Workshop Trends in Radio Resource Management, Barcelona, 16 November 2005.

Accepted but not already published Papers

Six papers have been accepted for presentation at the following international conferences.

- R. Ferrús, A. Gelonch, F. Casadevall, X. Revés, Nima Nafisi, “EVEREST Testbed: QoS Management Evaluation in B3G Networks”, 2nd International IEEE/Create-Net Conference on Testbeds and Research Infrastructures for the Development of Networks and Communities (TRIDENCOM 2006) to be held in Barcelona (Spain); 1-3 March 2006.
- J. Pérez-Romero, O. Sallent, R. Agustí, L. Wang, H. Aghavmi, “Network-Controlled Cell-Breathing for Capacity Improvement in Heterogeneous CDMA/TDMA Scenarios”, IEEE Wireless Communications and Networking Conference WCNC’06 to be held in Las Vegas (USA) 3-6 April 2006
- 4 papers at IEEE Vehicular Technology Conference (VTC’06 Spring) to be held in Melbourne (Australia) 7-10 May 2006.
 - J. Pérez-Romero, O. Sallent, R. Agustí, L. Wang, H. Aghavmi, “A Novel Algorithm for Radio Access Technology Selection in Heterogeneous B3G networks” .
 - J. Sánchez-González, J. Pérez-Romero, O. Sallent “A Location-Aware Resource Reservation Algorithm with User Class Differentiation in WCDMA”.
 - F. Adelantado, O. Sallent, J. Pérez-Romero, “On Modelling Spatial Traffic and Service Non-Uniformities in WCDMA Reverse Link”.
 - J. Pérez-Romero, O. Sallent, R. Agustí, “A Novel Framework for Robust WCDMA Planning under Changing Spatial Traffic Distributions”

Annex 2.- Relation with the Standards

As a matter of fact, the EVEREST project addresses topics such as Radio Resource Management (RRM), Common Radio Resource Management (CRRM) and QoS Management that are typically outside the scope of the Standardization fora, being mainly implementation dependent issues. However, as it is explained below, the EVEREST consortium has prepared some contributions reflecting the main outcomes of the project. These contributions have already been presented to the TSG RAN WG2 (Radio Layer 2 specification and Radio Layer 3 RRC specification) of the 3GPP or are pending of submission.

Impact of Current or Emerging Standards on Project's Work

The project has worked in close contact with the relevant standards bodies currently addressing the aspects treated by EVEREST. Therefore the project acquired on a regular basis useful documentation and information from standards bodies (3GPP, IETF mainly), regarding the Radio Resource management issues and the QoS handling. This has ensured that the project evolution has been always aligned with the guidelines provided by the standardisation framework.

Impact of Actual or Expected Contributions on the Work of Standard Bodies

The following papers and contributions have been presented at different standardisation groups:

- "UTRAN as a Part of a Multi-access Network", tdoc: REV-WS008: Source: TeliaSonera, 3GPP RAN Future Evolution Workshop, 2-3 November 2004, Toronto, Canada
- "Radio Resource Management Architecture considerations for Heterogeneous Wireless Systems - The EVEREST consortium" tdoc number R3-051250, Source: TeliaSonera, 3GPP RAN WG3 meeting 49 in Seoul (Korea), 7-11 November 2005.

Moreover 3GPP TSG RAN WG2 is currently developing a Technical Report TR 25.922: "Examples of RRM strategies for HSDPA. EVEREST project has prepared one contribution, reflecting the main outcomes of the project in terms of schedulers for HSDPA, at the 3GPP RAN WG2 meeting to be held in Denver (USA), 13-17 February 20062.